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Controlling and Monitoring Voice Quality in Internet Communication

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Controlling and Monitoring Voice Quality in Internet Communication

by

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of the requirements for the degree of
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DEDICATION

To my teachers, parents, wife, and children

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ABSTRACT

The Voice over Internet Protocol (VoIP) is on its way to surpassing toll quality. Although VoIP shares its transmission channel with other communication traffic, today internet has a wider bandwidth than the legacy Digital Loop Carrier and voice could be digitized higher than traditional 8 kbps, to say 16 kbps. Thus, VoIP should not be limited by the toll quality. However, VoIP quality could go down, as a result of unpredictable traffic congestion and network imperfections. These two situations cause delay jitter and packet loss of VoIP. To overcome these challenges, there are ongoing works for service providers including but not limited to optimizing routing and adding more bandwidth. There are also works by developers at the user's end, which includes compressing voice packet size and processing playout delay adapted to the network condition.

While VoIP planning or off-line quality monitoring and control use overall quality measurements such as mean opinion score (MOS) or R-factor, the real-time quality supervision typically uses the network condition factors only. The control mechanism that is based on network quality could adjust the channel parameter by changing Codec and its parameters, and changing playout delay, etc. to minimize the loss of voice quality.

As bandwidth plays a prominent role in IP traffic congestion, compressing the packet header is a possible solution to minimize congestion. Replacing a completed packet header with a smaller header will significantly reduce the packet header size. For instance, with a context, a compressed header will not consist of RTP header and, thus, could reduce 16 bytes from each packet. However, the primary question is how to deal with delay jitter calculation without time

stamping. In this research, a delay jitter calculation for VoIP packet without timestamp has been provided.

Compressing payload or using high compressing Codecs, is another major solution for preventing quality downgrade with limited bandwidth. The challenge with many Codec and the tradeoff between Codec quality and packet loss due to limited bandwidth has been addressed in this research with a summary of Codec quality evaluation and a bandwidth planning calculation.

Although the E-model and its R-factor has been proposed by the International Telecommunication Union (ITU) for VoIP quality measurement, with many network and Codec parameters, it could only be used for offline quality control. Since accessing a live traffic for monitoring live quality is somewhat impossible, at the client side, only packet loss and delay jitter matters. In this research, more in-depth investigation of adaptive playout delay based on jitter prediction has been carried out and recommended as the end user solution for quality improvement. An adaptive playout delay based on Markov model also has been developed in detail and tested with real VoIP network. This development has closed the gap between research and engineering. Therefore, the Markov model could be evaluated and implemented.

CHAPTER 1: INTRODUCTION

1.1 Background

Today the advent of network convergence has made it possible for the telephone, data and video services to be carried over in one network, the internet. The VoIP is on its way to replace the legacy telephony system [1-4]. Although VoIP has a positive potential to surpassing toll quality, it is always a concern for any service provider as well as the client application developer. Compressing packet size and optimizing playout delay are among the efforts to improve the voice quality. However, compressing packet by using higher compressing Codec could degrade the voice quality. Therefore, testing Codecs quality over VoIP [5,6] has been done widely. Compressing packet header [7] has also gone through extensive academic research.

Planning communication bandwidth and optimizing packet size to mitigate the impact of packet loss has become ubiquitous [7]. During these works, some quality evaluation methods have been developed. The quality measurements include both objective and subjective. While the subjective measure is only used for quality evaluation, the objective measurement such as delay jitter and packet loss ratio could be employed for quality control.

In the situation outside of what has been planned, such as impaired wireless communication or network roaming, where no dedicated channel or bandwidth could be assigned to VoIP channel, the only chance to limit the quality degradation is having a good packet loss conceal and adaptive jitter playout delay mechanism at the end-user side. Many studies had been carried out to improve

this opportunity [8,9]. Therefore, among state of the art studies, delay jitter prediction and playout delay have been addressed [10].

1.2 Motivation

The network planning is the first step of VoIP quality assurance and calculating the required bandwidth is the first task for VoIP planning. The research objective is to make it straightforward and familiar, from some previous proposals and suggestions [1,7].

Using Codec is the only method for reducing payload. However, using Codec will also reduce voice quality. How to evaluate a speech Codec and which Codec could be used for VoIP is the question for any VoIP research.

On the other hand, reducing the Internet packet header size is one possible solution to minimizing bandwidth [7]. For instance, removing UTP/RTP header could cut 20 bytes from each packet [8,9]. However, one primary question is how to deal with delay jitter calculation without time stamping.

Playout delay is the only one solution for the end-user for reducing packet loss caused by delay jitter. Having an extended playout delay could minimize the packet loss ratio. However, this will degrade the voice quality. Optimized or adaptive playout delay should be an important feature for VoIP usage. The state of the art jitter prediction based on Markov model [11-12] has been studied by others for adaptive playout delay control application. However, many questions such as whether Markov model is a practical method, how to implement it and what is the quality have still not been answered yet. Therefore, more studies need to be carried out in order to respond to these questions.

Due to all these reasons, our research consists of the following four tasks:

- Task #1: Planning a minimum bandwidth for a VoIP channel.

- Task #2: Evaluation of Speech Codecs.
- Task #3: Calculation of delay jitter without timestamp.
- Task #4: Continue pending research work on Markov model for delay jitter prediction.

1.3 Research Contributions

From the requirements for quality assessment of VoIP, our research provides a summary of Codec quality assessment and how to calculate the bandwidth planning for a VoIP channel, for a variable Codec frame length and variable bit rate that others have not mentioned before.

The research has proposed a calculation method for jitter delay without timestamp. This work eliminates the doubt that jitter cannot be found without timestamps and it allows the developer to implement header compression while still being able to measure the delay jitter.

We have built a mathematical model based on Markov's theory and other works by others. The model uses quantized jitter as model states. We found that the Markov model will have problems if a jitter state is not present in the model. This may be the reason why the Model has not been further developed. We have provided a solution to overcome the infinite calculations of the Model that lack a jitter state. Then we continued our research by testing the feasibility and precision of the playout delay method based on the Markov model with an actual network, and with different model steps. We have concluded that this approach is useful, and how to use it in the best possible way. We also compared the Markov method with other methods to confirm the accuracy and simplicity of our approach.

1.4 Dissertation Structure

Chapter 1 provides a general introduction to VoIP, motivations, and the contributions of this dissertation. Chapter 2 introduces the fundamentals of VoIP, including voice process, IP stack,

switching strategy and voice coding. It also describes the analysis on header compression and provides a brief review of the VoIP architecture and protocol.

Chapter 3 describes speech Codec for VoIP application as well as Codecs quality evaluation methods. Chapter 3 also discusses how to improve VoIP quality during the planning stage and the trade-off between Codec and bandwidth.

Chapter 4 provides a review of VoIP quality measurement and how to reduce the impact of the network impairment, mainly focusing on delay jitter issues. Chapter 4 also presents a summary of research on jitter measurement without packet timestamps and an adaptive playout delay based on Markov chain model with quantized delay jitter. Some tests have been introduced and experimental results are presented in this chapter. A brief discussion on Kalman filter [13-14] and Maxwell model for packet loss is provided.

Chapter 5 provides a summary and suggestions for future research work.

CHAPTER 2: VOICE OVER INTERNET PROTOCOL

2.1 Overview

Voice-over-Internet-Protocol (VoIP) [1] is the technique used to carry voice signal over an IP network. In VoIP, the voice signal is segmented into frames and stored in voice packets. The voice packet is transmitted using IP in compliance with one of transmitting multimedia format (voice, video, fax, and data) across a network protocol, i. e., H.323 (ITU), MGCP (level 3, Bellcore, Cisco, Nortel), SIP (IETF), IAX2 (Digium), MEGACO/H.GCP (IETF), T.38 (ITU), SIGTRAN (IETF), Skinny (Cisco), etc. As a typical communication network, VoIP is composed of three basic parts: switching, terminal, and transmission systems. However, the VoIP transmission system is borrowed from another communication network: The Internet. VoIP is a staking-up protocol from Internet Protocol. Typical Internet applications use TCP/IP, in addition, VoIP uses RTP/UDP/IP. Although IP is a connectionless effort network communication protocol, TCP is a reliable transport protocol that uses acknowledgment and retransmission to ensure packet receipt. Used together, TCP/IP is a reliable connection-oriented network protocol suite. VoIP term is also known as IP telephony. VoIP is used as a substitution of legacy telephony.

A large number of factors impact VoIP quality. However, network impairment is the most dominant factor that could cause the loss, delay and delay jitter of packets, which in the end will reduce the VoIP quality. In this chapter, we will summarize current academic research on VoIP technique and analyze the factors that influence VoIP quality.

2.2 VoIP Network

2.2.1 Review of the Layered Structure of TCP/IP Family

While the Internet protocol (IP) deals only with packets, Transmission Control Protocol (TCP) will allow two hosts to establish a connection and send and receive streams of data. TCP guarantees delivery of data and also guarantees that packets will be delivered in the same order in which they were sent.

IP represents one component of the TCP/IP (Transmission Control Protocol/ Internet Protocol) family [15,16]. It is difficult to discuss IP as a separate entity unto itself. The TCP is a session layer protocol. The TCP coordinates the transmission, reception, and retransmission of packets in a data network to ensure reliable communication. The TCP protocol also coordinates the division of data information into packets. The TCP will add sequence and flow control information to the packets, confirm packets that are lost during a communication session. TCP utilizes IP as the network layer protocol.

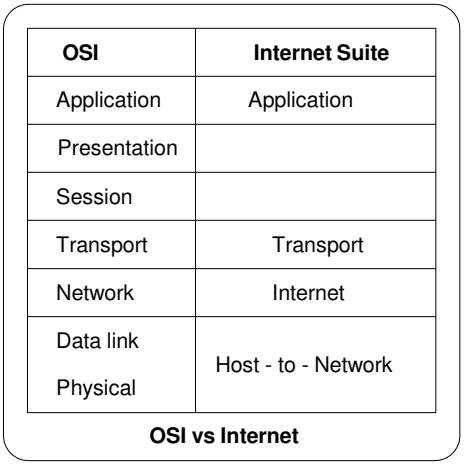


Figure 1 The seven layers of OSI model is stacked into four layers of TCP/IP

The Open Systems Interconnection (OSI) model has been used widely by networks as a reference. The OSI reference model was developed as a mechanism to subdivide networking function into logical groups of related activities referred to as layers. Due to the complexity of the

seven layers model, other simpler model such as the Internet is used. Figure 1 illustrates how to stack the seven layers of OSI model into four layers of TCP/IP.

A simple VoIP protocol architecture is illustrated in Figure 2. The stack provides Real-time Transport Protocol (RTP), User Datagram Protocol (UDP), call-setup signaling (i.e., H.323, SIP) and QoS feedback RTP Control Protocol (RTCP) [17].

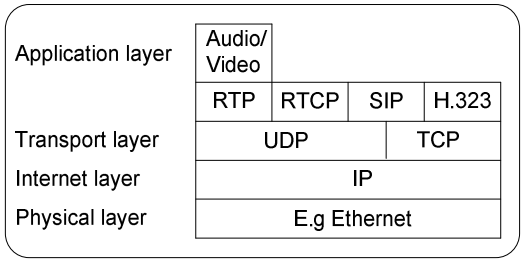


Figure 2 VoIP Protocol stack

2.2.2 Review of the Layered IP Stack

A basic Voice over IP packet contains a header and a payload as shown in Figure 3. The header will be constructed as follows:

MAC header	IP header	UDP header	RTP message

Figure 3 VoIP header

Whereas the IP header is 20 bytes for IP version 4 (Figure 4) or 40 bytes for version 6, UPD header is 8 bytes, and RTP is 12 bytes long. The total is 40 bytes. Each part of VoIP packet is described as follows:

00	**01**	**02**	**03**	**04**	**05**	**06**	**07**	**08**	**09**	**10**	**11**	**12**	**13**	**14**	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**	**28**	**29**	**30**	**31**
Version				IHL				TOS								Total length															
Identification																Flags			Fragment offset											
TTL								Protocol								Header checksum															
Source IP address																															
Destination IP address																															
Options and padding :::																															

Figure 4 IP header (version 4)

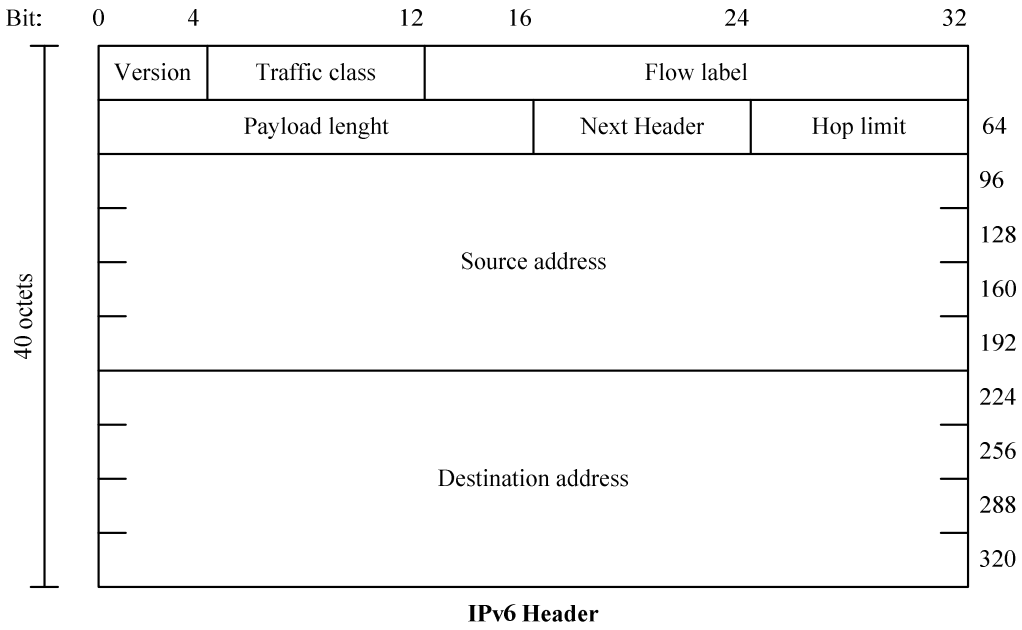


Figure 5 IP header (version 6)

Figures 4 and 5 illustrate IP headers byte chart. First 20 bytes are mandatory. Figure 6 is optional UPD header, and Figure 7 is the RTP header.

00	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
Source Port																Destination Port															
Length																Checksum															
Data :::																															

Figure 6 UDP header (version 4)

00	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
Ver		P	X	CC				M	PT							Sequence Number															
Timestamp																															
SSRC																															
CSRC [0..15] :::																															

Figure 7 RTP header (version 4)

The VoIP header length takes up VoIP traffic significantly. For an instant, if each packet contains 10 ms voice segment (100 packets/sec), there are 100 headers per second, a minimum 32 kbps of bandwidth will be required for just headers transmission [18].

2.2.3 VoIP Payload

VoIP packet consists of a header as described above and a payload. Payload carries voice information (in-band) or signal (out-band). If it is voice, it will be a Codec segment. The purpose of the Codec is to reduce the payload, thus reducing the transmission bandwidth. Using Codecs is one of the reasons for degrading voice quality. More about Codec and Codec evaluation will be discussed in Chapter 3.

2.2.4 Header Compression

Header compression [19] has been used to reduce transmission bandwidth by reducing packet size. The header compression works on a context by creating a context identifier (CID) at the beginning of each flow. The header will be compressed by the compressor after the context is established on both sides, and appends the CID at the transmittal end. The decompressor decompresses all the header by using the CID to refer to the context at the receiver end.

In the case of header information remaining the same for difference packets, the header compression seems very helpful in reducing bandwidth [7,19]. The measurement of delay jitter on a packet that has UDP/RTP header removed was done and is described in Chapter 4.

2.2.5 VoIP Architecture

Figure 8 illustrates a “hybrid” VoIP network, in which the VoIP is not staying isolated. VoIP is still able to reach out to a legacy voice client, i.e., analog telephones and facsimiles, vice versa. A gateway is an interfacing device between a non-IP and an IP client. A network address translator will be used for a voice channel that passes through different IP networks (i.e., LAN-WAN). An in-band or out-band signaling payload will control the interface between non-IP and IP client.

A cellular phone will be served by the nearest cellular station, which today is a part of IP network. On the other side, most PBX also has IP trunk along with legacy analog/TDM trunks. An application could turn any "smart" device that has built-in microphone and speakerphone into a voice client.

2.2.6 Soft-switch; Signaling and Payload Transport

Today all network switches are soft switches. Switching a voice packet would be the same as switching any other IP packet. The control and synchronize signaling of a voice call, i.e., ring, transfer, could be sent as a special payload and will be generated and detected by context that is defined by the application. The conventional payload will carry the voice.

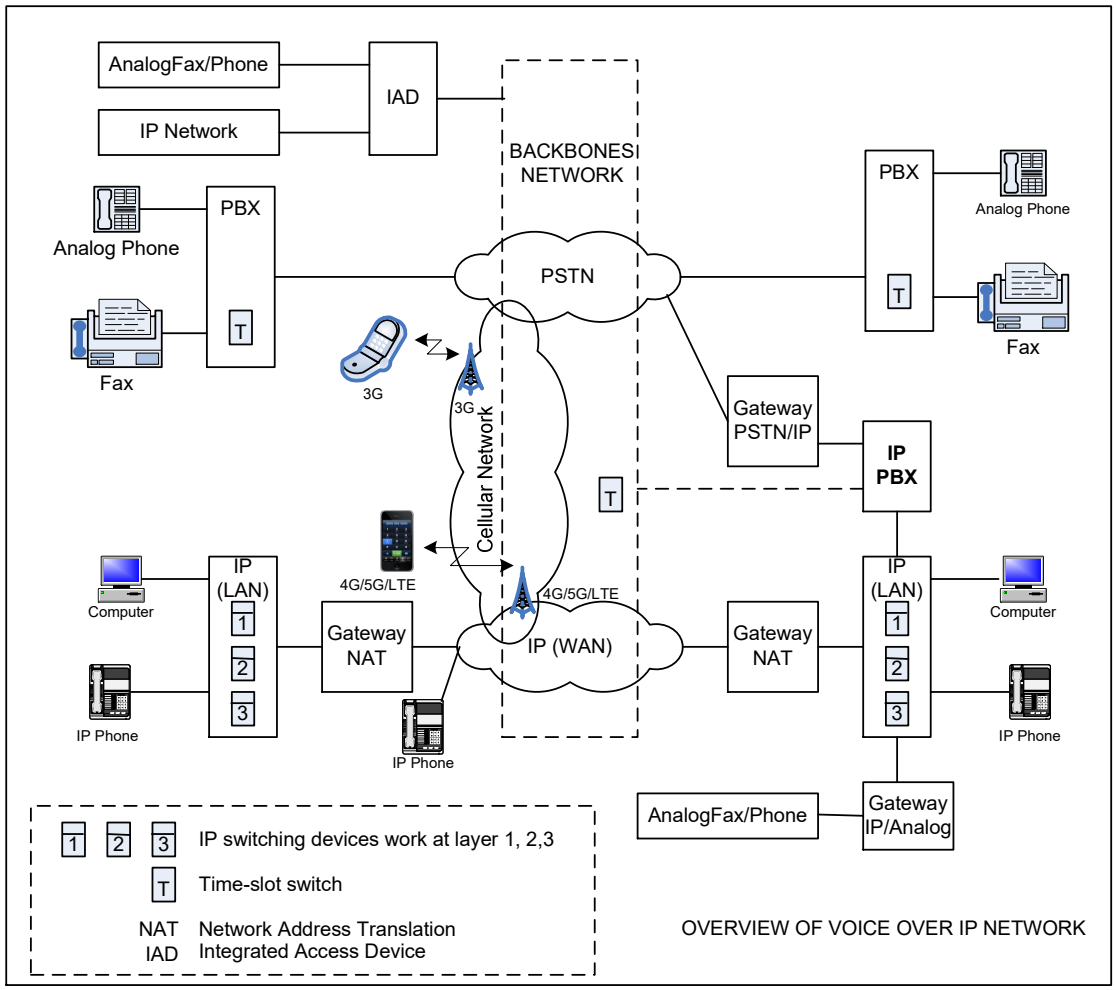


Figure 8 Overview of VoIP network

2.2.7 VoIP in 4G/5G and LTE Communication

The cellular network has become a part of IP network. Each client (cell phone) is an integrated smart device. It can work at any communication layer. Application (app) could make a mobile device work for multi-subscribers. Nonetheless, the legacy phone conversation is just one of the device application. The signaling protocol could be different with each subscriber. However, the voice packetizing method remains the same via a data network. Along with legacy phone call, we can make a voice call by using many types of internet messaging systems [20-22], i.e., Skype, Viber, WhatsApp, etc.

2.2.8 VoIP with IPv6

Internet Protocol version 6 (IPv6) has addressed the issue of IP address in IPv4. IPv6 also consist of eight bits traffic class in IP header. The traffic class could be used to identify the class of service, a solution for QoS control if another packet should have a lower class of service. In IPv4 the class of service will be identified by the priority of traffic port in UDP header. IPv6 also has 20 “label flow” bytes which have not been standardized for use yet. Since VoIP header for IPv6 is 80 bytes, double of IPv4, and no quality improvement has been proven yet, today VoIP over IPv6 is still limited in small deployment or for evaluation purpose only [23-25].

2.2.9 Summary

VoIP consist of an extended header and a payload. VoIP packet should be compressed to reduce the bandwidth. There are two possible processes which could result in reducing bandwidth, compressing the header and/or payload. Compressing the header could cause losing some real time information. Compressing the payload could reduce voice quality. There are challenges of recovering the effects of real-time data loss and minimizing voice quality degradation.

VoIP with IPv6 is still under development and evaluation because IPv6 standard has not been finalized yet. One of the issues with IPv6 on VoIP is its header consists of 80 bytes. That takes up more bandwidth than IPv4.

CHAPTER 3: SPEECH CODEC AND EVALUATION AND SELECTION OF SPEECH CODEC FOR VOIP APPLICATION

3.1 Overview

This chapter describes speech Codec for VoIP application as well as Codecs quality evaluation methods. A discussion on how to improve VoIP quality during the planning stage and the trade-off between Codec and bandwidth also are provided.

3.2 Codec

A speech Codec is a hardware device or software program that is capable of converting an analog voice into digital data stream and back. Also, some speech Codecs use compression techniques that remove redundant information, by replacing a long real bit stream with a small coded stream. The purpose of compression is to reduce the size of bit stream needed to encode the information, thereby, reducing the amount of time or bandwidth required for transmission.

3.2.1 Classification of Speech Codecs

Speech coding [5,6,26] schemes are primarily classified as waveform coding and parametric coding or vocoding. A derivative of the above coding classes is hybrid coding which combines waveform and parametric coding techniques. Waveform speech coders encode an original speech waveform in the time domain or frequency domain at a given bit-rate. The recovered audio signal on the decoder side is an approximate replica of the original sound. In waveform coding, the original sound characteristics are present at the output of the coder and, as such, the process is termed as a non-perceptual process. In contrast to waveform coder, vocoder encodes voice based on parameters that characterize individual sound segments. Typically, the

decoder reconstructs a new and often different waveform that will have a similar sound. This difference is the reason why vocoders are also known as parametric coders. In vocoding, the original sound represented by the extracted parameters at the output of the coder is termed as a perceptual process. Despite needing longer segments, vocoders operate at lower bit rates than waveform coders, but the reproduced speech quality usually suffers from a loss of naturalness and the characteristics of an individual speaker. Such distortions caused by the modeling inaccuracy are often very difficult to remove. Finally, hybrid speech coder is one that borrows some features from vocoders, even though it belongs to the family of waveform coders.

3.2.2 Analog – Digital Conversion

The traditional A/D conversion allows an analog signal to be transported via the digital channel. Coding is a process in which an analog signal (voice or speech) is transformed into a digital signal. Decoding is a process in which the digital signal is converted back to an analog signal. Two critical parameters of an A/D conversion are the sampling frequency or the sampling rate (samples/second) and the quantization resolution or word length (bits/sample). The bit rate is the product of these two values. Lower bit rate results in higher compression. Higher compression is achieved by reducing the sampling rate and/or the word length. Lowering the sampling rate means reducing the time resolution, however, the lowest sampling frequency is limited by the Nyquist theorem [27]. On the other hand, reducing the word length lowers the amplitude resolution or increases the quantization error. Typical A/D conversion allows setting the quality to be nearly perfect, i.e., very high bit-rate. Depending on the class of service, a sub-coding process could be used to re-sample digital speech to a smaller bit-rate. The A/D conversion is a process at both ends of “mouth to ear” path, where the analog signals are converted to digital and reconverted back to analog by a Digital to Analog (D/A) conversion.

3.2.3 Waveform CODEC

In waveform coding, an analog signal is digitized without requiring any knowledge of how the signal was produced. A waveform coder attempts to mimic the waveform as closely as possible by transmitting the actual time or frequency domain magnitudes.

Among waveform coders are Pulse Code Modulation (PCM), Differential PCM (DPCM), adaptive DPCM (ADPCM) [28], Adaptive Predictive Coding (APC) [29], Delta Modulation (DM) [30], Subband Coding (SBC) [31], and Adaptive Transform Coding (ATC) [32]. The PCM is a most commonly used waveform coding technique, which is based on a three-step process: Sampling, Quantization, and Encoding.

- Sampling: Typically, the analog speech signal is sampled at 8000 samples/sec.
- Quantization: In quantization process, the sampled signal amplitudes are assigned values from a pre-defined set of quantized amplitudes. The difference between the adjacent quantized values represents the step size (granularity) of the quantizer. Most of the speech quantizers use 8-bit binary code to represent a sample. However, the step size used for encoding signals may not be uniform. Non-uniform quantization is used because there is a higher probability of occurrence of lower peak-to-peak signals than higher peak-to-peak signals. Most of the PCM systems today use companding process, followed by uniform quantization to reduce the numbers of bits necessary to encode each PCM sample to 8 bits.

Figure 9 illustrates a simplest waveform-coding scheme. An analog signal sample is taken at every cycle of sampling impulse by the sampler. The quantizer compares samples with a pre-defined scale and gives an output with a number of the bit as the desired word length. The coder

then sends out the impulse as bit by bit from quantizer output. The coding clock frequency is equal to the word length times sampling frequency.

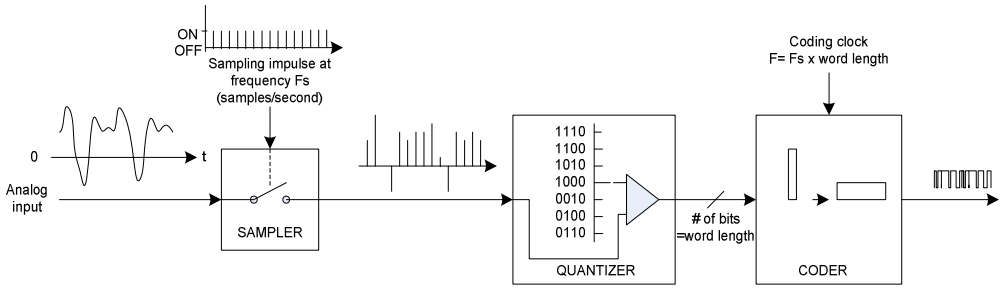


Figure 9 A simplest waveform-coding scheme

The A-law companding (used in Europe) and μ-law companding (used in North America) are two ways to compress and decompress PCM voice data [33]. The behavior of A-law companding is depicted by equation (1). Per ITU G.711, each PCM word consists of three parts as shown in Figure 10. The first bit is the polarity bit, the next three bits represent chord number, and the remaining four bits represent one of 16 possible steps within a chord. Chords are spaced logarithmically, whereas steps within the chord are linearly spaced.

$$Y = \begin{cases} \frac{A_x}{(1+logA)}; & 0 \le \frac{V}{A} \\ \frac{1+\log(A_x)}{(1+logA}; & \frac{V}{A} \le v \le V \end{cases} \quad (1)$$

where v represents the instantaneous input amplitude, A is a constant set to 87.56, and V represents the maximum input amplitude.

The behavior of μ-law companding is represented by equation (2):

$$\mu = \frac{\log(1+\mu x)}{\log(1+\mu)} \quad (2)$$

where μ has a constant value of 255 and x has value of v/V and varies between -1 and 1.

Usually the A/D IC (Integrated Circuit) consists of a built-in PCM hardcode with A-law and μ-law selection [34]. In practice, these three processing steps (sampling, quantization and encoding) could take place simultaneously.

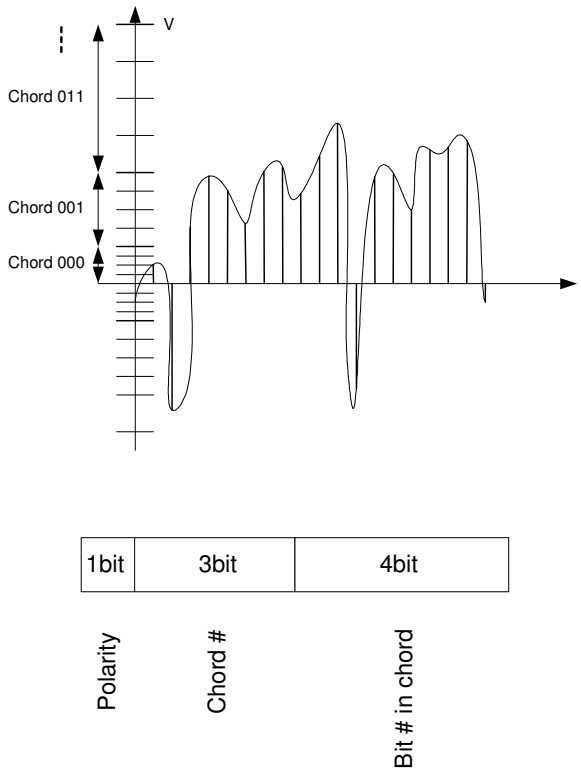


Figure 10 PCM coding and PCM word

3.2.4 Voice Codec or Vocoder

Today's techniques for speech synthesis and recognition are based on the model of human speech production. By looking at the characteristic of the human voice over a short segment (10-20 ms), it either sounds as voiced or fricative (unvoiced). Voiced sounds occur when the air is forced from the lungs through the vocal cords and out of the mouth and nose. During that, vocal cords vibrate at frequencies between 50 to 1000 Hz, resulting in periodic puffs of air being injected into the throat. Vowels are an example of voiced sounds. Fricative sounds occur when the air flow is nearly blocked by the tongue, lips, or teeth, resulting in air turbulence near the constriction. Fricative sounds include f, sh, z, v, etc. [34].

In Vocoder Coding, a short segment (10-20 ms) of the human voice as described could be produced or classified as voiced (i.e., /a/, /e/) or unvoiced (i.e., /sh/,/w/). Voiced sounds are

represented by the periodic excitation with the pitch (i.e., fundamental frequencies) being an adjustable parameter. On the other hand, an unvoiced sound is more like a random noise generator. Figure 11 illustrates the general speech production model employed by the vocoder. The vocoder design deals with three major issues, namely, quality, bitrate, and processing power.

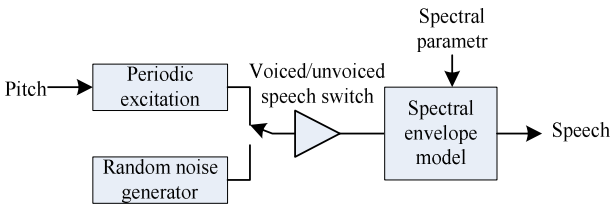


Figure 11 Vocoder block diagram

Several different vocoders have been developed in the market. Among of them, Homomorphic and Linear Predictive Vocoders (LPV) [35] are the most popular. The LPV is the most useful method for a quality speech coding at a very low bit rate. The LPV computes the coefficients of the filter to minimize the error between the prediction and the actual sample.

3.2.5 Hybrid Codec

Hybrid coding is a compromised solution between the high quality of Waveform coding and the synthetic quality of Vocoder. The key difference between Linear Predictive Coding (LPC) vocoder and LPC hybrid is the method of modeling speech. LPC-based vocoder uses a model that concentrates on voiced and unvoiced portions of speech, and with analysis-by-synthesis hybrid coder. The selection of an excitation signal compensates for the residue problem. The well-known hybrid coder families are RPE-LPC (Regular Pulse Excitation LPC), MPE-LPC (Multi-Pulse Excited LPC), and CELP (Code-Excited Linear Predication) [36].

3.2.5.1 Regular Pulse Exited Coding

RPE-LPC is the coding method used for the Global System for Mobile Communication (GSM). The GSM full rate speech Codec operates at 13 kbps and uses a Regular Pulse Excited

(RPE) Codec [35]. In the RPE, the length of speech frame segment is 20 ms long, and each frame contains a set of eight short-term predictor coefficients. Each frame is then further split into four 5 ms sub-frames, and for each sub-frame, a delay and gain for the Codec's long-term predictor will be decided by the encoder. The residual signal after both short and long term filtering is quantized for each sub-frame [37]. The residual signal of forty samples is decimated into three possible excitation sequences, each consisting of 13 samples. The best representation of the excitation sequence and each pulse in the sequence has its amplitude quantized with three bits which will be chosen by the sequence with the highest energy.

At the decoder, the reconstructed excitation signal is fed through the long-term and the short-term synthesis filters to give the reconstructed speech. A post filter is used to improve the perceptual quality of this reconstructed speech. The GSM Codec provides good quality speech, although not as good as slightly higher rate G728 Codec. However, the main advantage of GSM Codec over other low rate Codecs is its relative simplicity.

The RPE-LPC GSM representative is GSM 06.10.

3.2.5.2 Multi Pulse Excited Coding

Figure 12 shows the block diagram of an LPC speech synthesizer with multi-pulse excitation (MPE-LPC). Compared with the traditional LPC synthesizer, MPE-LPC doesn't have the pulse and white noise generators and the voiced-unvoiced switch. The excitation for the all-pole filter is generated by an excitation generator that produces a sequence of pulses located at times $t_1, t_2, \ldots t_n \ldots$ with amplitudes $a_1, a_1, \ldots a_n \ldots$, respectively. If desired, a pole-zero filter could replace the all-pole filter. The sampled output of the all-pole filter is passed through a low-pass filter to produce a continuous speech waveform S_i.

In MPE-LPC, pulse position is found by an exhaustive search based on minimized mean squared error as shown in Figure 13 [38].

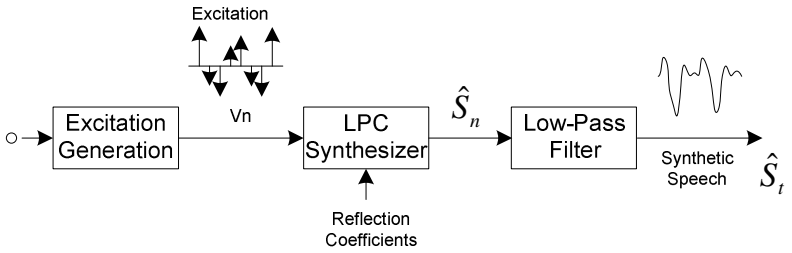


Figure 12 LPC speech synthesizer with multi pulse excitation

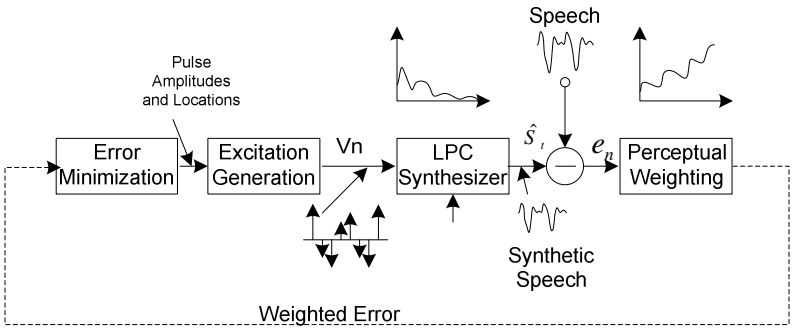


Figure 13 Analysis-by-synthesis procedure for the multi-pulse excitation

Due to its synthetic quality, MPE-LPC is no longer very popular.

3.2.5.3 Code Excited Linear Predictor (CELP) Coders

CELP [39] employs both waveform and vocoding techniques. In CELP, speech is passed through a vocal tract and pitch predictor, an index from codebook will be used in place of an actual quantization of the excitation signal (see Figures 14 and 15). The data rate of CELP is between 4.8 and 16 kbps. Some versions of CELP are listed below:

- FS 1016: Data rate is 4.8 kbps. It is the U.S Department of Defense standard.
- The G.728 Recommendation: An ITU standard, operates at 16 kbps, and provides toll-quality speech comparable to the 32 kbps ADPCM.
- The G.729 Recommendation: An ITU standard, operates at 8 kbps. Due to the complexity of G.729, several annexes are written for G.729.

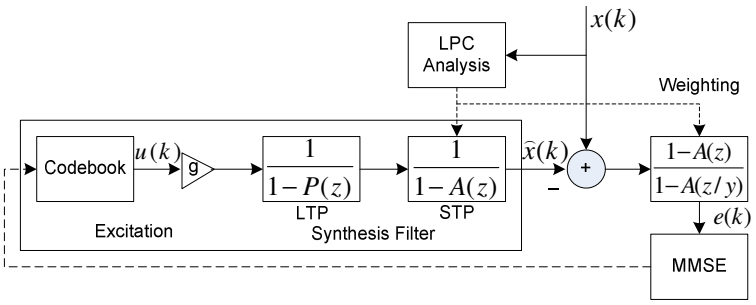


Figure 14 CELP encoder

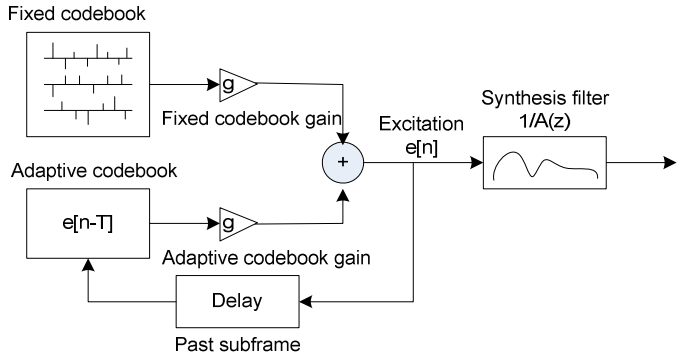


Figure 15 CELP decoder

- The G.723.1 Recommendation: An ITU standard coder, operates at 5.3 and 6.3 kbps.
- Vector sum excited linear prediction (VSELP), a speech coding method used in several cellular standards, including IS-54 and IS-136 (2G mobile phone system). The VSELP algorithm is known as an analysis-by-synthesis coding technique. It belongs to the class of CELP.
- Algebraic Code Excited Linear Prediction (ACELP) is patented by VoiceAge Corporation. It has a limited set of pulses which is distributed as excitation to linear prediction filter. The representatives of ACELP are GSM 06.20 Half-Rate (HR) and GSM 06.60 Enhanced Full Rate (EFR).

3.2.6 Other Vocoders

Since different Codecs could be implemented on the same hardware platform; there are several other free-of-charge Codecs which have been developed by the open source community as the alternative for licensed Codecs, utilizing the power of the open source community.

3.2.6.1 Internet Low Bit-rate Codec (iLBC)

The iLBC is a VoIP Codec created by Global IP Sound. iLBC (internet Low Bit-rate Codec) is a free speech Codec suitable for robust voice communication over IP [26]. iLBC is designed for narrow band speech. It has a payload bit rate of 13.33 kbps with an encoding frame length of 30 ms. It also has a bit rate of 15.20 kbps with an encoding length of 20 ms. This Codec is equipped with graceful speech quality degradation in the case of lost frames, which occur in connection with lost or delayed IP packets [39]. Global IP sound's aim is for iLBC to have a basic quality and robustness to packet loss higher than G.729A, and the computational complexity similar to G.729A.

3.2.6.2 GIPS

Originally, GIPS [40] was also created by Global IP Sound. The owner claims to be able to maintain voice quality even with 30% packet loss. GIPS is the technology licensed for use by Skype. It is being made an IETF standard. GIPS operate at bit rates of 13.3 kbps and up. GIPS wideband Codecs (16 kHz sample rate) include:

- iSAC: Internet Speech Audio Codec is a high-efficiency variable bit rate Codec. iSAC is targeted for low data rate connections including dialup. It most closely matches the one described as being used by the Skype client.
- iPCM-wb: Internet Pulse Code Modulation wide-band for higher rate connections.

3.2.6.3 Speex

Speex [26,40] is an Open Source/Free Software patent-free designed for speech. Per Speex Project team, it is free of charge to lower the barrier of entry for voice applications. Speex is well-adapted to Internet applications. It also provides useful features that are not present in most of the other Codecs. Today, Speex is part of the GNU Project and is available under the Xiph.org variant of the BSD license. Speex is a great Codec due to its flexibility. However, it is also an expensive Codec since it consumes more CPU power than the G729, G726 or GSM Codecs, and just about the same as iLBC.

3.2.6.4 LPC-10

The LPC-10 Codec derives its name because it uses 10 LP coefficients. The LPC-10 operates at a bit rate of 2.4 kbps and with a total of 54 bits per frame. LPC-10 is used for narrow bandwidth connections. The disadvantages of using LPC-10 are [41] listed below:

- Decoded voice can sound very “buzzy” which is caused by parameter updates.
- Poor LP modeling results in wide bandwidths and rapid decay of the pulse excitation.
- Regularly voiced excitation is unnatural - normally some jitter.
- Voicing errors produce significant distortions.
- Binary voicing decision is sometimes poor.
- Not suited to model nasals - although okay in practice.
- Only models speech – does not work if background noise exists (i.e., not suited to mobile phone applications without further work).

3.2.7 Media Format Codecs

Media format high-quality Codecs, such as MP3 (MPEG audio layer III), AAC (Advanced Audio Codec), WMA (Windows Media Audio), Ogg Vorbis, etc… are used in Audio storage, i.e., CD, Television, DVD, Blue-ray, camcorder, etc… Due to one or more of the following reasons such as high complexity, high bit rate, and long delay, media format Codecs have not been used for real-time VoIP conversation. However, they could be used for music on hold or recorded announcements playback. Vorbis Codec is a free and open source Codec. Its quality is comparable with other commercial Codecs (MP3, WMA, AAC…). Typical of media format Codec for music Vorbis Codec have a bit rate of 128 kbps. Encoding and decoding delay times are not revealed, in fact from seconds to minutes. Even if media Codecs are used for music on hold application, the playback bit rate will be very low. We shall, therefore, concentrate on the Speech Codec, and will not have further discussion on media format Codecs.

3.2.8 Codec Loss Concealment Algorithm

In order to reduce the impact of frame loss [42], some Codecs such as G.729, G.723.1, AMR and the iLBC have a built-in loss concealment algorithm. The loss concealment algorithm can interpolate the parameters for the loss frames from the parameters of previous frames. For example, in the G.729 Codec, the loss concealment algorithm repeats the line spectral pair coefficients of the last good frame. The adaptive and fixed codebook gain will be taken from the previous frames. However, they are damped to reduce their impact gradually. The fixed codebook contribution will be set to zero if the last reconstructed frame was classified as voiced, The pitch delay is taken from the previous frame and is repeated for each of the following frames. The adaptive codebook contribution will be set to zero, and the fixed codebook vector will be randomly chosen if the last reconstructed frame was classified as unvoiced. In other words, if a frame is not

losing all parameters, it will be re-constructed based on received and previous parameters instead of replacing the whole frame with interleaving frame, which is the previous reconstructed frame.

3.3 Evaluation of Speech Codecs

In general, the performance of speech and audio Codecs is evaluated using six attributes: *bit rate, speech quality, signal delay, complexity, robustness to acoustic noise, and robustness to channel errors*. The desired Codec must have low bit rate, low delay, less complexity, but high speech quality. Speech quality can be determined both subjectively and objectively.

3.3.1 Subjective Measures

Subjective measurements are obtained from the listening tests, whereas objective measurements are computed directly from the coded speech parameters. Some common subjective measures are listed below:

- *Diagnostic Rhyme Test (DRT):* It uses a set of isolated words to test for consonant intelligibility in initial position. The DRT is one of the ANSI S3.2-2009 standards for measuring the intelligibility of speech over communication systems.
- *Paired Comparison Test (PCT)*: Pair comparison method is usually used to test the overall system acceptance. It is based on a speech synthesizer listener which will listen to artificial speech for hours per day [43-44]. Stimuli from each synthesizer will be compared in pairs with all n(n-1)/2 combinations. If there are more than one test sentence (m), each version of a sentence will be compared to all the other version. Thus there will be a total number of n(n-1)m/2 comparison pairs.
- *Mean Opinion Score (MOS)*: The listener's task is simply to evaluate the tested speech with scale described in Table 1. In Unified Communication (UC), there are two classes of MOS, listening quality (MOS-LQ) and conversional quality (MOS-

CQ). Another MOS scale, is known as the DMOS (Degradation MOS) or the DCR (Degradation Category Rating) and it is an impairment grading scale to measure how the different disturbances in speech signal are perceived (Absolute Category Rating).

Table 1 Scales used in MOS and DMOS

RATING	MOS (ACR)	DMOS (DCR)
5	Excellent	Inaudible
4	Good	Audible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

Calls made over the PSTN have a MOS score of around 4.3, while the vocoders used in wireless telephone system, i.e., GSM (Global System for Mobile Communication), CDMA (Code Division Multiple Access) and TDMS (Time-Division Multiplexing System) have MOS score ranging from 3.4 to 3.9. The subjective measures give a wide variation among listener scores since the scales used by the listeners are not calibrated and do not provide an absolute measure. In VoIP application, subjective measures do not indicate specific network impairment, which is important for VoIP quality control. The objective measures indicate multiple factors, including network impairment status, and therefore has been widely used in VoIP control and monitoring.

3.3.2 Objective Measures

H. Özer et al. [5] categorize the objective measures into *perceptual* and *non-perceptual* groups. The non-perceptual group is further divided into time-domain and frequency-domain measures. The metrics used in time domain measure of speech quality includes Segmental Signal-to-Noise Ratio (SNRseg), Signal-to-Noise Ratio (a special case of SNRseg), Czenakowski Distance (CZD). The metrics used in frequency domain measure of speech quality includes Log-Likelihood Ratio (LLR), Log Area Ratio (LAR), Itakura-Satio Distance measure (IS or ISD),

COSH Distance measure (COSH), Cepstral Distance Measure (CDM), Spectral Phase (SP), Spectral Phase-Magnitude distortion (SPM), and Short Time Fourier-Radon Transform measure (STFRT). The perceptual group of speech quality measure includes Barker Spectral Distortion (BSD), Modified Barker Spectral Distortion (MBSD), Enhanced Modified Barker Spectral Distortion (EMBSD), Perceptual Audio Quality Measure (PAQM), Perceptual Speech Quality Measure (PSQM), Weighted Slope Spectral Distance Measure (WSSD), and Measuring Normalizing Blocks (MNB). A select set of above-mentioned measures calculate distortion from the overall data, namely, SNR, CZD, SP and SPM. On the other hand, the distortion is calculated for small segments and then the average is taken over all the segments to obtain the overall speech quality measure. The measures using the averaging include SNRseg, BSD, MBSD, EBSD, PAQM, PSQM, LLR, LAR, ISD, COSH, CDM, and WSSD. The segment length is 20 ms (320 samples for 16 kHz signal), which is used as window size for the techniques MNBs and STFRT.

Another way to classify objective measure is intrusive or non-intrusive. Intrusive or non-intrusive measures relate to voice quality measurement over the network. Intrusive methods are more accurate but are usually unsuitable for monitoring live traffic because of the need for reference data and access to the network. Current non-intrusive methods rely on subjective tests to derive model parameters. Therefore these methods are limited and do not meet new and emerging applications.

3.3.2.1 Time-Domain Measures

Time-domain measures compare the two waveforms – the original audio signal, *x(i)* and the recovered audio signal, *y(i)* in the time domain. Some popular time-domain measures are:

Segmental Signal-to-Noise Ratio (SNRseg) is defined in equation (3) as the average of the SNR values over small segments:

$$SNRseg = \frac{10}{M}\sum_{m=0}^{M-1}\log\left(\sum_{i=Nm}^{Nm+n-1}\left(\frac{x^2(i)}{(x(i)-y(i))^2}\right)\right) \quad (3)$$

The length of the segment is typically 15 to 20 ms for speech. The SNRseg is applied to frames with energy above a specified threshold in order to avoid silence regions.

Signal-to-Noise Ratio (SNR) in equation (4) is a special case of SNRseg, when M=1 and one segment encompasses the whole record. The SNR is very sensitive to the time alignment of the original and the distorted audio signal. The SNR is measured as:

$$SNR = 10\log\left(\frac{\sum_{i=1}^{N}x^2(i)}{\sum_{i=1}^{N}(x(i)-y(i))^2}\right) \quad (4)$$

This measure has been criticized for being a poor estimator of subjective audio quality.

Czenakowski Distance (CZD) is a correlation-based metric, which directly compares the time-domain sample vectors as shown by equation (5):

$$C = \frac{1}{N}\sum_{i-1}^{N}\left(1 - \frac{2*\min(x(1),y(i))}{x(i)+y(i)}\right) \quad (5)$$

3.3.2.2 Frequency-Domain Measures

Frequency-domain measures (e.g. LLR, LAR, ISD, COSH, CDM, WSSD, SPD, SPMD, STFRT) [5] compare the original and recovered signals on the basis of their spectra or in terms of a linear model based on second order statistics [45].

Log-Likelihood Ratio (LLR), also known as Itakura distance, considers an all-pole linear predictive coding (LPC) model of the speech segment, $x(n) = \sum_{m=1}^{p} a(m)x(n-m) + G\,u(n)$ where, *a(m)* are the prediction coefficients, p is the filter order, and *u(n)* is an appropriate excitation source. The LLR measure is then defined by equation (6):

$$LLR = \log\left(\frac{a_x^T R_y a_x}{a_y^T R_y a_y}\right) \quad (6)$$

where a_x is the LPC coefficient vector for the original signal *x(n),* a_y is the corresponding vector for the recovered signal *y(n),* with respective covariance matrix R_y.

Log Area Ratio (LAR) is another LPC-based technique, which uses partial correlation (parcor) coefficients. The parcor coefficients form a parameter set derived from the short-time LPC representation of the speech signal under test. The LAR will be delivered from area ratio functions of these coefficients as equation (7):

$$LAR_i = \log\left(\frac{A}{A_{i+1}}\right) = \log\left(\frac{1+\alpha_1}{1-\alpha_1}\right), \quad A_{p+1} = 1 \tag{7}$$

where α_i is the i[th] parcor coefficient, which can be found by using equation (8):

$$\alpha = \alpha_i^{(1)} \quad , 1 \le i \le p \tag{8}$$

where $\alpha_i^{(1)}$ is the i[th] LPC calculated by using the i[th] order LPC model.

Itakura-Saito Distance Measure (ISD) is the discrepancy between the power spectrum of the recovered signal *Y(w)* and that of the original audio signal *X(w)*:

$$IS = \frac{1}{2\pi}\int_{-\pi}^{\pi}\left(\frac{X(w)}{Y(w)} - \log\frac{X(w)}{Y(w)} - 1\right)dw \tag{9}$$

COSH Distance Measure is the symmetric version of the ISD. Here the overall measure is calculated by averaging the COSH values over the small segments:

$$COSH = \frac{1}{2\pi}\int_{-\pi}^{\pi}\left[\frac{1}{2}\left\{\left(\frac{X(w)}{Y(w)} + \frac{Y(w)}{X(w)}\right) - \log\left(\frac{X(w)}{Y(w)} + \frac{Y(w)}{X(w)}\right) - 2\right\}\right]dw \tag{10}$$

Cepstral Distance Measure (CDM) is a distance, defined between the cepstral coefficients of the original and recovered signals. The cepstral coefficients can also be computed by using LPC parameters. An audio quality measure for the m[th] frame based on the L cepstral coefficients, $c_x(k)$ and $c_y(k)$, of the original and recovered signals respectively, is given by equation (11a):

$$d(c_x, c_y, m) = \left(\left[c_x - c_y(0)\right]^2 + 2\sum_{k=1}^{L}\left[c_x(k) - c_y(k)\right]^2\right)^{1/2} \tag{11a}$$

The overall distortion is calculated over all frames using equation (11b).

$$CD = \frac{\sum_{m=1}^{M} w(m) d(c_x, c_y, m)}{\sum_{m=1}^{M} w(m)} \tag{11b}$$

where M is the total number of frames, and $w(m)$ is a weight associated with the m^{th} frame.

For example, the weighting could be the energy in the reference frame. It is typical to use a 20 ms frame length and the energy of the frame as weights.

In Spectral Phase and Spectral Phase-Magnitude Distortions, the phase and/or magnitude spectrum differences have been observed to be sensitive to image and data hiding artifacts. They are defined by equations (12) and (13).

$$SP = \frac{1}{N}\left(\sum_{w=1}^{N} \left|\theta_x(w) - \theta_y(w)\right|^2\right) \tag{12}$$

$$SPM = \frac{1}{N}\left(\lambda \sum_{w=1}^{N} \left|\theta_x(w) - \theta_y(w)\right|^2 + (1-\lambda) \sum_{w=1}^{N} ||X(w) - |Y(w)||^2\right) \tag{13}$$

where SP is the spectral phase distortion, SPM is the spectral phase-magnitude distortion, $\theta_x(w)$ is the phase spectrum of the original signal, and $\theta_y(w)$ is the phase spectrum of the distorted signal, $X(w)$ is the magnitude spectrum of the original signal, $Y(w)$ is magnitude spectrum of the distorted signal, and λ is chosen to attach commensurate weights to the phase and magnitude terms.

Short-Time Fourier-Radon Transform Measure (STFRT) is a multi-dimensional measure, based on Short-Time Fourier Transform (STFT). Given a Short-Time Fourier transform (STFT) of a signal, its time projection provides the magnitude spectrum while its frequency projection yields the magnitude of the signal itself. By considering all the other dimensions rather than taking only the vertical and horizontal projections, the Radon transform of the STFT measure could be obtained. STFRT is the objective audio quality measure based on the mean-square distance of Radon transforms of the STFT of two signals.

3.3.2.3 Perceptual Measures

Perceptual measures, such as WSSD, BSD, MBSD, EMBSD, PAQM, PSQM, and MNB, take explicitly into account the properties of the human auditory system [5].

Bark Spectral Distortion (BSD) is assuming that speech quality is directly related to speech loudness. The BSD estimates the overall distortion based on the average Euclidian distance between loudness vectors of the original and the distorted audio. The Bark spectral distortion in [45] is calculated using equation (14) as shown below:

$$BSD = \sum_{i=1}^{K} \left[S_x(i) - S_y(i) \right]^2 \quad (14)$$

where K is the number of critical bands, $S_x(i)$ is the Bark spectra of the ith critical band corresponding to the original, and $S_y(i)$ is the coded speech.

For speech, 18 critical bands (which is up to 3.7 kHz) are used. The overall distortion will be calculated based on averaging the BSD values.

Modified Bark Spectral Distortion (MBSD) is a modification of the BSD. MBSD incorporates noise-masking threshold to differentiate between audible and inaudible distortions. The inaudible loudness difference, which is proportional to $S_x(i) - S_y(i)$ and below the noise-masking threshold will be excluded in the calculation of the perceptual distortion. The perceptual distortion of the nth frame is the sum of the loudness difference which is greater than the noise masking threshold as shown on following equation (15) as:

$$MSBD = \sum_{i=1}^{K} M(i) D_{xy}(i) \quad (15)$$

where *M(i)* denote the indicator of perceptible distortion and $D_{xy}(i)$ is the loudness difference in the i^{th} critical band, and K is the number of critical bands.

The global MBSD value will be calculated by averaging the MBSD scores over non-silence frames [5].

Enhanced Modified Bark Spectral Distortion (EMBSD) is a variation of MBSD. In EMBSD, only the first 15 loudness components (instead of the 24-Bark bands) will be used to calculate loudness differences. Loudness vector is normalized, and a new cognition model will be assumed based on post-masking effects as well as temporal masking.

In Perceptual Audio Quality Measure (PAQM), a model for emulating the human auditory system will be used. The transformation from the physical to the psychophysical domain is performed by time-frequency spreading and level compression, for example masking behavior of the human auditory system is taken into account. In the beginning, the reference and coded signals are transformed into short-time Fourier domain (Figure 16), then the frequency scale will be converted into pitch scale (in bark) and the signal will be filtered to transfer from outer ear to inner ear. These results will be in the power-time-pitch representation. Therefore, the resulting signals will have frequency domain smearing and time domain smearing. Per Thilo Thield and Ersnt Kabot of Technical University of Berlin and others, the measure of the quality of an audio system is an average of comparison.

Perceptual Speech Quality Measure (PSQM) was devised by Beerends in 1993. This development represents an adapted version of the more general perceptual audio quality measure (PAQM), which is optimized for telephony speech signals. PSQM is a modified version of the PAQM [45], in fact, the optimized version for speech. PSQM does not include temporal or spectral masking for loudness computation. PSQM applies a nonlinear scaling factor to the loudness vector of distorted speech. PSQM has been adopted as the ITU-T Recommendation P.861, its detailed block diagram shown in Figure 17 which illustrates how to calculate PSQM. The P.861is end-of-life, its successor, is P.682 – Perceptual Evaluation of Speech Quality (PESQ). Our research has no intention to develop any test using PSQM.

The Perceptual Evaluation of Speech Quality (PESQ) model begins by a standard listening level aligning both signals, then modeling a standard handset by filtering (using an FFT) with an input filter. The signals are then processed through an auditory transform which is similar to that of PSQM. At this process, there is also an equalizing for linear filtering and for gain variation. Two distortion parameters will be extracted from the disturbance, and will be aggregated into frequency and time, and will be mapped to a prediction of subjective MOS.

The PESQ aims to have more suitability with the nowadays network, especially VoIP, in comparison with previous models, i.e., PSQM, BSD, etc., PESQ has better performance to deal with prediction accuracy, taking proper account of noise or packet loss, delay jitter, etc.

Weighted Slope Spectral Distance Measure (WSSD) uses a filter bank [46], consisting of thirty-six overlapping filters of progressively larger bandwidth which can make short-time audio spectrum smoother. The filter bandwidths approximate critical bands in order to give equal perceptual weight to each band. Klatt [47-48] uses weighted differences between the spectral slopes in each band because the spectral variation could play a major role in human perception of audio quality. The spectral slope is computed in each critical band as:

$$V_x(k) = X(k+1) - X(k) \tag{16a}$$

$$V_y(k) = Y(k+1) - Y(k) \tag{16b}$$

where k is the critical band index, $X(k)$ and $Y(k)$ are the spectra in decibels, and $\{V_x(k), V_y(k)\}$ are the first order slopes of these spectra.

Next, a weight for each band is calculated based on the magnitude of the spectrum in that band as shown in equation (17).

$$WSSD = \sum_{k=1}^{36} w(k)[V_x(k) - V_y(k)]^2 \tag{17}$$

where, the weight $w(m)$ is chosen according to a spectral maximum.

WSSD is computed separately for each 12 ms audio segment and then by averaging the overall distance.

Measuring Normalizing Blocks (MNB) is an objective speech measure that provides an algorithmic estimate for rating human subjects that will give coded or degraded speech [49]. It is based on a model of human auditory perception and has been optimized against a large number of human-rated speech passages. In MNB the important role of the cognition module for estimating speech quality has been emphasized. MNB is sensitive to the relative delay between the reference and the test signals. The human listeners' sensitivity to the distribution of distortion is considered in MNB, so MNB uses hierarchical structures that have a time and frequency scales from larger to smaller. MNB integrates over frequency scales. It measures differences over time intervals. It also integrates over time intervals, and it measures differences over frequency scales. These MNBs is then linearly combined to estimate overall speech distortion.

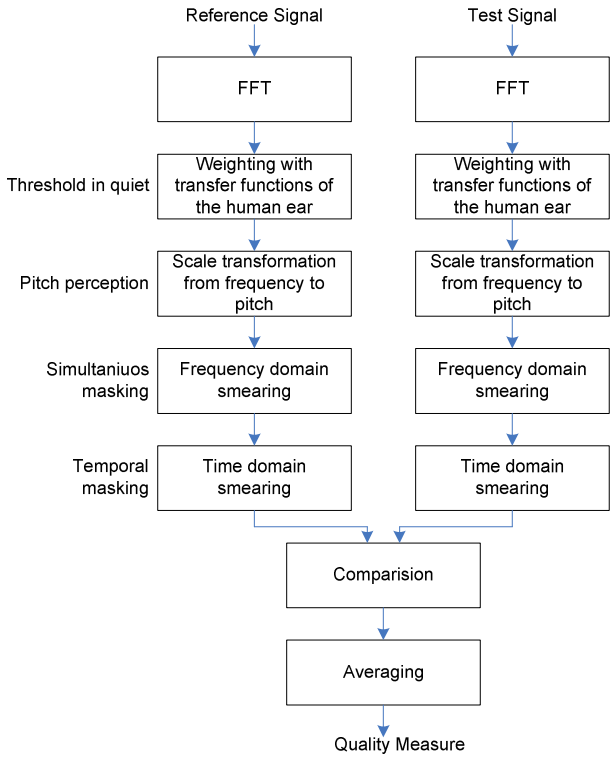


Figure 16 Perceptual Audio Quality Measure (PAQM)

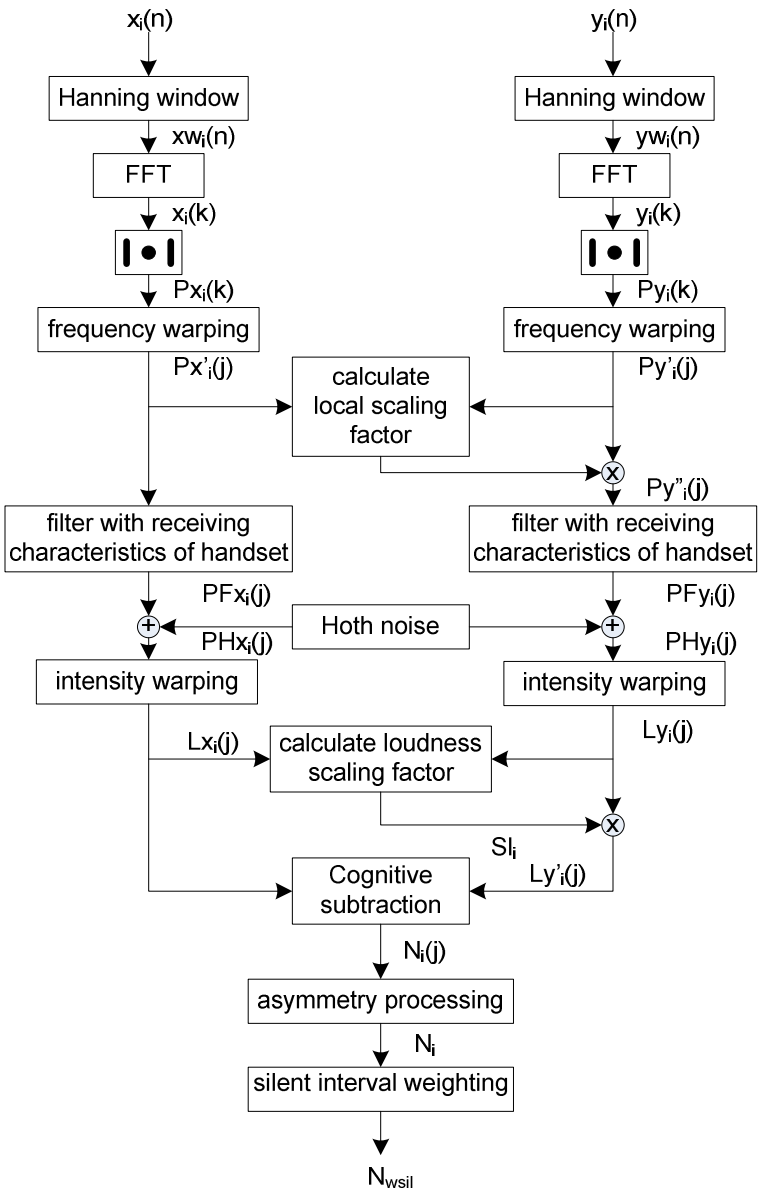


Figure 17 PSQM calculation procedure

3.4 Objective Quality Measures Evaluation

In this research, we have performed all of the tests for the object quality measures mentioned in section 3.3.2 and compared with the listening evaluation. We found that all objective measures performances are not linear with MOS. Each objective measure result depends on the language and background noise. All measures were performed with off-line samples with no network impact. The final voice quality of a VoIP channel, in fact, depends on many other factors,

included but not limited to network quality. The voice quality of a VoIP channel will be discussed in following sections. In conclusion, PESQ is selected as the preferred method for speech Codec evaluation, due to its accuracy and its simplicity.

3.5 Selection of Speech Codecs

While legacy public switched telephone network (PSTN) has a dedicated medium for voice transmission, VoIP uses Internet service medium for transmission. In addition, VoIP stream is always carried out using packetized form (packet voice) which has an IP header. As a result, VoIP has a higher delay and higher packet loss probability. The selection of Codec for VoIP depends on the network quality and Codec specification. The lower bit-rate Codec is preferred for low bandwidth service; the small packet is preferred for long delay network. Selecting a Codec is also based on Processor power versus Codec complexity.

Variable Bit Rate (VBR) Codec has been developed, however, the bit rate change was based on the speech property itself. The Codec selection or bit rate could be changed based on network condition or by class of service that is paid by the client. Emmanuel Antwi-Boasiako et al. [26] has performed a test on two popular Codecs, G.711 and Speex with objective Perceptual Evaluation of Speech Quality (PESQ) and subjective Mean Opinion Score (MOS). The report did not mention whether a narrow band or wide band Speex has been used for testing. There is other research on voice quality and bandwidth tradeoff and voice Codec for a specific language.

Among the number of Codecs for VoIP, the following are our rating for the Codecs from the highest to the lowest:

- Speex for its flexibility, quality and low implementation cost, no license fee. Especially with 16 kbps Speex quality is better than G.711's.
- G.729 for its low bandwidth and good quality.

- G.711, Annex 1, higher bandwidth, good quality and good packet loss recovery, no license fee.
- G.723 for the lowest bandwidth.
- iLBC, for open source VoIP application.

3.5.1 Codec Impairment

In ITU G.107 recommendation for VoIP transmission planning, E-model is used to calculate the transmission rating factor R.

E-model is looking at both IP and non-IP factors which impact on VoIP QoS. Codec impairment is represented as $I_{e\text{-}eff}$ in the E-Model, which is described in ITU G.113 (Table I.1/ITU G.113). More details of E-model by ITU and also mentioned by E. Myakotnykh [50] in his dissertation will be presented in Chapter 4.

3.5.2 Codec vs. Bandwidth

Using Codec with low $I_{e\text{-}eff}$ [51] is desirable. However, the Codec may not be selected based on provisioning $I_{e\text{-}eff}$ value only. Low compression and long interval Codec can cause higher bandwidth, more delay, congestion, or packet loss. Codec selection is based on available bandwidth. Using a Variable Bit Rate (VBR) Codec could improve the quality, however for bandwidth planning a fixed bit rate will be used for calculation.

Most of the service providers will assist their customer based on bandwidth requirements. From an academic standpoint, a simple calculation of the required bandwidth for a VoIP channel has been proposed.

Assuming that minimum bandwidth B is requested to assure that there is no packet loss caused by queuing delay, R is maximum Codec rate (bps), n is the number of packets per second $n = 1/T_s$, where T_s is minimum Codec frame length.

Assuming that H is the header size (usually 40 bytes = 320 bits), Then minimum bandwidth for VoIP is:

$$B = nH+R = R+H/T_s \quad (18)$$

Below are two examples for G.729A and G.711:

(a) With G.729A, R=8 kbps, T_s=0.02 s, B = 8000+320/0.01 = 24 kbps.

(b) With G.711, R=64 kbps, T_s=0.02 s, B = 76 kbps.

Note that since the Variable Bit Rate (VBR) could be used, we use maximum Codec bit rate and minimum frame length which haven't been mention by B. Goode [1] or B. Ngamwongwattana [7] or others before. The frame length T_s typical is 20 ms and always less than preferred maximum delay of 400 ms. Additional bandwidth may be required for signaling (out-band signal). For reference, Table 2 provides the bandwidth requirement by Cisco [52].

Table 2 Provisional planning values for the equipment impairment factor Ie per ITU G.113

Codec Information				Bandwidth Calculations					
Codec & Bit Rate (Kbps)	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	MOS	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Packets Per Second (PPS)	Bandwidth MP or FRF.12 (Kbps)	Bandwidth w/cRTP MP or FRF.12 (Kbps)	Bandwidth Ethernet (Kbps)
G.711 (64 Kbps)	80	10	4.1	160	20	50	82.8 Kbps	67.6 Kbps	87.2 Kbps
G.729 (8 Kbps)	10	10	3.92	20	20	50	26.8 Kbps	11.6 Kbps	31.2 Kbps
G.723.1 (6.3 Kbps)	24	30	3.9	24	30	34	18.9 Kbps	8.8 Kbps	21.9 Kbps
G.723.1 (5.3 Kbps)	20	30	3.8	20	30	34	17.9 Kbps	7.7 Kbps	20.8 Kbps
G.726 (32 Kbps)	20	5	3.85	80	20	50	50.8 Kbps	35.6 Kbps	55.2 Kbps
G.726 (24 Kbps)	15	5	-	60	20	50	42.8 Kbps	27.6 Kbps	47.2 Kbps
G.728 (16 Kbps)	10	5	3.61	60	30	34	28.5 Kbps	18.4 Kbps	31.5 Kbps

3.5.3 Codec vs. Complexity

Even though Codec processor power (Million instructions per second - MIPS) is not a concern for engineers nowadays, typical encoding and decoding will be carried out by terminal devices (phone) without any problem (DSP is capable of hundreds of MIPS). However, in channel encoding and decoding, wherever traffic load is high, the cost for high MIPS implementation is an issue. Variable Bit Rate (VBR) is desired to improve VoIP QoS, VBR Codec is more complicated than CBR (Constant Bit Rate) Codec. However, nowadays the cost of very powerful DSP is negligible, and therefore the complexity is no longer an issue.

3.5.4 Codec Selection Based on Implementation Cost

Choosing a Codec sometimes depends on the license fee. Most of VoIP servers using open source are also using open source or license free Codecs such as G.711, GSM or Speex.

3.6 Speech Codec Summary and Future Challenges

VoIP is a real-time packet communication. Packet loss and delay jitter are two major concerns when selecting a Codec. Language oriented Codec could be an approach for VoIP Codecs. Applying noise and echo cancellation before compressing the speech is always recommended [53]. In addition, a Codec with other features, such as accent, language recognition, could reduce the bit rate without reducing quality.

Speech Codec is used for packetizing in VoIP. Codec could use different compressing techniques to reduce the bandwidth. Speech Codecs are specified as Waveform Codec, Vocoders, and Hybrid coders. Codec quality evaluation could be objective or subjective. With the subjective method, it could be intrusive or non-intrusive models. Codec has a dominant impact on the VoIP quality. Selecting a Codec for VoIP is based on planning, including the provision of Codec impairment, allocating bandwidth and service class. Using more complex Codecs or a Codec

translator can generate a significant delay. Future challenges of speech Codec work include achieving lower bit-rate but higher quality, improvement in loss concealment and reduced complexity.

CHAPTER 4: QUALITY CONTROL AND IMPROVEMENT

4.1 Overview

The VoIP perceived Quality of Service (QoS) is dependent on equipment impairment and network quality as described in Figure 18. All Codec measures described in previous chapters are used to measure the quality of speech Codec, which indicates the Codec impairment level only. Voice over network quality depends on many factors including Codec quality and network quality. It is necessary to have an objective measurement or prediction model, which includes all factors that influence voice over network quality [54].

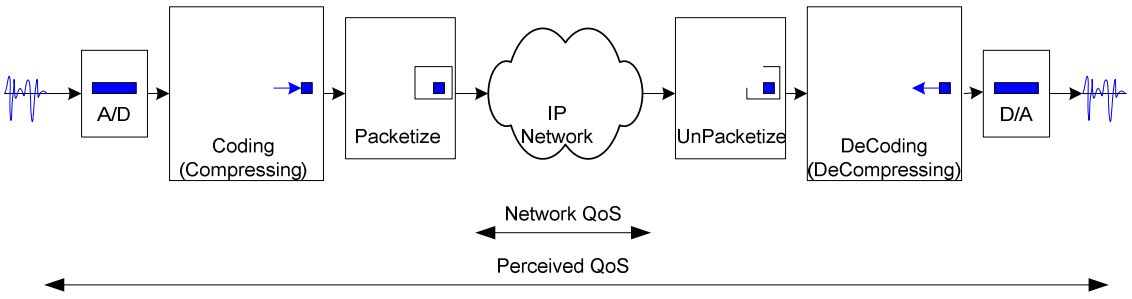


Figure 18 Perceived QoS zone

VoIP quality measurement includes subjective and objective methods. MOS is the well-known subjective method while E-model is the most popular objective method.

Packet loss and delay jitter are two major network impairment factors that impact the VoIP quality. Packet loss could be caused by delay jitter. Having a playout delay could reduce the packet loss. However, a long delay could reduce the voice quality as well. Therefore optimizing playout delay has been addressed. In this chapter, delay jitter will be measured even when there is no packet timestamp. Delay jitter will be quantized and used for a Markov chain prediction and will

be used to control playout delay time. Some of the experiment results will be provided to validate this model. Other possible filters for jitter prediction such as Kalman filter [14] and packet loss modeling that have been recently addressed will also be discussed.

4.2 Subjective Measurement

As mentioned in the previous chapter, Mean Opinion of Score *(MOS),* the subjective measure used in voice communication is the most widely used and simplest method to evaluate speech quality in general. The subjective measures give a wide variation among listener scores since the scales used by the listeners are not calibrated and do not provide an absolute measure [55]. In VoIP application [7], subjective measures do not indicate specific network impairment, which is necessary for VoIP quality control. The objective measures indicate multiple factors [56], including network impairment status, and this is the reason why it is widely used in VoIP control and monitoring.

4.3 Objective Measurement

VoIP quality is dependent on the IP network and the end-point process quality. However, subjective measures do not indicate specific network impairment, which is important for VoIP quality control. Our goal is to provide a VoIP QoS strategy that allows monitoring and planning the QoS through the network from end to end, which includes:

- QoS of voice stream through the gateway.
- QoS of voice stream over local area network (LAN).
- QoS of voice stream over wide area network (WAN).

The strategy is to use objective measures that indicate multiple factors, including network impairment status, which has been used widely in VoIP control and monitoring. The weight of each impairment factors will reflect on the quality factor R of E-Model that will be discussed

below. R-factor will not only help the VoIP provider make the best trade-off decision between latency (delay), jitter, echo, network congestion, packet loss, and arrival of packets in out-of-sequence but also will be able to advice the VoIP users as to what they should do to control and monitor the VoIP QoS at their end.

The R-factor was described in the ITU-T G.107 recommendation in the second half of 2004. It defines a computing model known as an *E-model* [50,57-58]. The R-factor is a well-tried tool for transmission planning and for determining the combined impact of various transmission parameters that influence the call quality. As shown in equation (19), all appropriate transmission parameters are put together to calculate the R-factor as follows:

$$R = R_O - I_S - I_D - I_{E\text{-}EFF} + A \tag{19}$$

where, R_O is the basic signal-to-noise ratio, I_S is impairment that occur simultaneously with speech (e.g. quantization noise, received speech and sidetone levels), I_D is impairment that is delayed with respect to speech (e.g. talker/listener echo and absolute delay).

$I_{E\text{-}EFF}$ captures effects of special equipment or equipment impairment (e.g. Codecs, packet loss and jitter), and A is an advantage factor (permitted range is from 0 to 20; 0 for wired line and 10 for GSM). A short form of (19) is:

$$R = R_0' - I_D - I_{E\text{-}EFF} \tag{20}$$

where R_0' is representing non-network impairment factors, which usually comes as a default value.

It is well known that the R depends on loss and delay jitter; these impairments will be represented by I_D and $I_{E\text{-}EFF}$.

The E-model is a non-intrusive voice quality prediction, however, it has a number of limitations. For example, it is based on a complex set of fixed, empirical formulas and is limited by the number of Codecs and network conditions (because subjective tests are required to derive

model parameters), which hinders its use in new and emerging applications. It is a static model which cannot adapt to the dynamic environment of IP networks [59] and based on the assumption that the individual impairment factors defined on the transmission rating scale are independent of each other, which may not be true. R model, however, is useful for estimating the QoS of a VoIP channel given static information, or a good measure for VoIP planning.

Table 3 Equipment impairment factor to bandwidth requirement for Codec (Source: Cisco)

Codec type	Reference	Operating rate kbit/s	Ie-value
PCM (see Note)	G.711	64	0
ADPCM	G.726, G.727	40	2
	G.721(1988), G.726, G.727	32	7
	G.726, G.727	24	25
	G.726, G.727	16	50
LD-CELP	G.728	16	7
		12.8	20
CS-ACELP	G.729	8	10
	G.729-A + VAD	8	11
VSELP	IS-54	8	20
ACELP	IS-641	7.4	10
QCELP	IS-96a	8	21
RCELP	IS-127	8	6
VSELP	Japanese PDC	6.7	24
RPE-LTP	GSM 06.10, Full-rate	13	20
VSELP	GSM 06.20, Half-rate	5.6	23
ACELP	GSM 06.60,Enhanced Full Rate	12.2	5
ACELP	G.723.1	5.3	19

ITU has made an online tool available for E-model computing at: http://www.itu.int/ITU-T/studygroups/com12/emodelv1/calcul.php (Figure 19) [60]. More information regarding to the E-Model can be found at ITU website http://www.itu.int.

For MOS friendly user, equation (21) could be used to convert R-factor to MOS [29,50]:

$$MOS = \begin{cases} 1 & R < 0 \\ 1 + 0.035R + R(R-60)(100-R)*7*10^{-6} & 0 < R < 100 \\ 4.5 & R > 0 \end{cases} \quad (21)$$

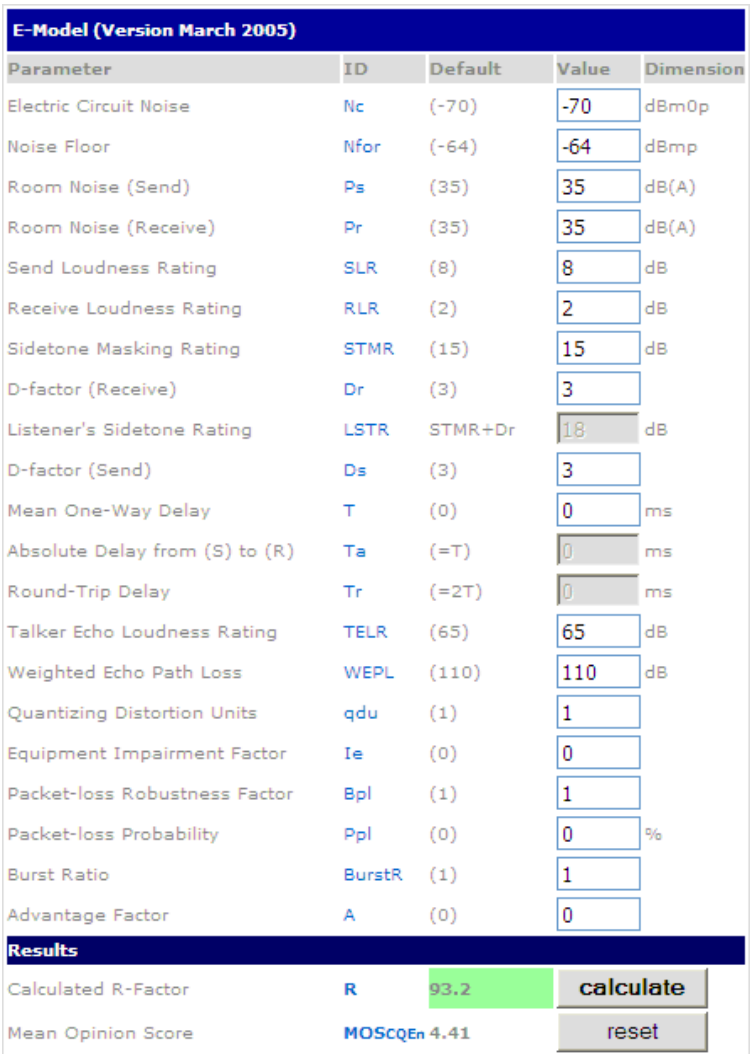
4.4 Latency, Delay Jitter, and Packet Loss

On the network side, the VoIP quality is dependent on these major factors: Latency, delay jitter and packet loss.

4.4.1 Latency

The delay time from "mouth to ear" of a VoIP channel include the following items:

- *Transmission Delay:* This delay depends on the speed or the data rate of the communication link and the packet length. It is the amount of time required to transmit all the packet's bits from the first bit to the last bit into the communication link, and this delay is proportional to the packet length.
- *Propagation Delay*: This delay depends on the physical characteristics of the communication link. Propagation delay is the time to transmit one bit over a link (i.e., the delay between the transmissions of the packet last bit from the source to the reception of last bit of the packet at the destination).
- *Switching Delay:* This is the time required to shift data packets through the various network hardware components such as hubs, routers. This delay is a reflection of the speed of the switching device, like transmission delay in a packet switch.
- *Queuing Delay*: This delay depends on the traffic on the communication link and capacity of switching device. Queuing delay is the delay between the entry point of a packet in the transmit queue to the actual transmission point of the message.
- *Processing Delay*: This delay depends on the speed of processor(s), load and type of processing scheme. This is the delay to process, compress and de-compress (Codec) data, and also other process such as echo cancellation.



E-Model (Version March 2005)

Parameter	ID	Default	Value	Dimension
Electric Circuit Noise	Nc	(-70)	-70	dBm0p
Noise Floor	Nfor	(-64)	-64	dBmp
Room Noise (Send)	Ps	(35)	35	dB(A)
Room Noise (Receive)	Pr	(35)	35	dB(A)
Send Loudness Rating	SLR	(8)	8	dB
Receive Loudness Rating	RLR	(2)	2	dB
Sidetone Masking Rating	STMR	(15)	15	dB
D-factor (Receive)	Dr	(3)	3	
Listener's Sidetone Rating	LSTR	STMR+Dr	18	dB
D-factor (Send)	Ds	(3)	3	
Mean One-Way Delay	T	(0)	0	ms
Absolute Delay from (S) to (R)	Ta	(=T)	0	ms
Round-Trip Delay	Tr	(=2T)	0	ms
Talker Echo Loudness Rating	TELR	(65)	65	dB
Weighted Echo Path Loss	WEPL	(110)	110	dB
Quantizing Distortion Units	qdu	(1)	1	
Equipment Impairment Factor	Ie	(0)	0	
Packet-loss Robustness Factor	Bpl	(1)	1	
Packet-loss Probability	Ppl	(0)	0	%
Burst Ratio	BurstR	(1)	1	
Advantage Factor	A	(0)	0	

Results

Calculated R-Factor	R	93.2	calculate
Mean Opinion Score	MOSCQEn	4.41	reset

Figure 19 An E-model calculation tool

Figure 20 illustrates total "mouth to ear" delay of a VoIP channel. The network delay is not a constant for each packet as the queuing times are not the same. Delay and loss are two major quality factors of a VoIP channel. Figure 21 illustrates the perceived voice quality based on network and application performances and channel noise, whereas each factor could result in both network and application performances.

- Network Performance:
  - Switching delay: Queuing and switching delay.
  - Propagation delay: The physical delay caused by physical propagation.

- Delay jitter.
- Packet loss caused by long delay or switching error.

- Application performance (at the client sides):
  - Codec delay and quality loss: Codec should wait until full voice frame length has completed and the digitization process reduce the voice quality.
  - Processing delay: Algorithmic and Codec delay due to Voice processing, coding and decoding.
  - Switching to/from network delay: The delay due to time to complete handing over or to receive one packet to the network.
  - Playout delay.
- Channel noise.

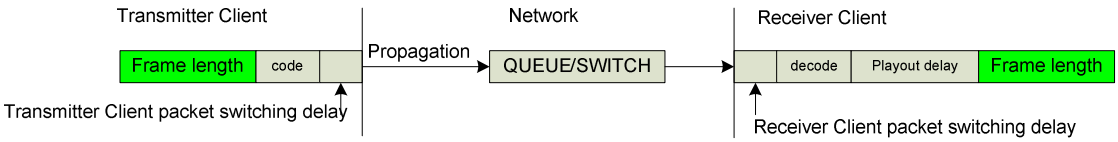


Figure 20 Total "mouth to ear" delay in VoIP

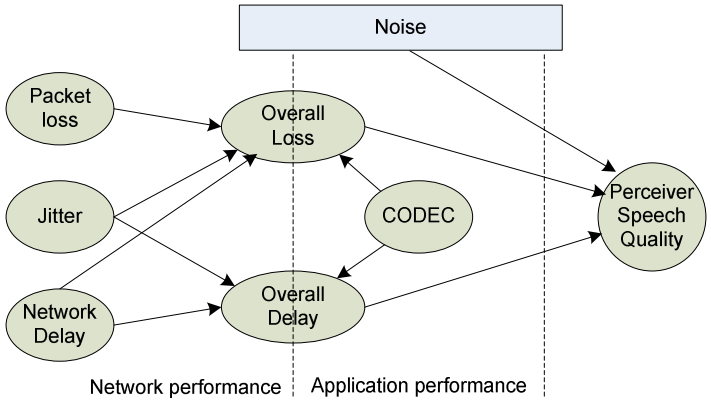


Figure 21 Perceived voice quality based on network and application performance and channel noise

4.4.2 Switching and Queuing

Figure 22 illustrates a simple and so-called NxN time-slot switch. A packet from input will be switched to the corresponding output. The congestion will occur when there is more than one packet to be switched to the same output at the same time.

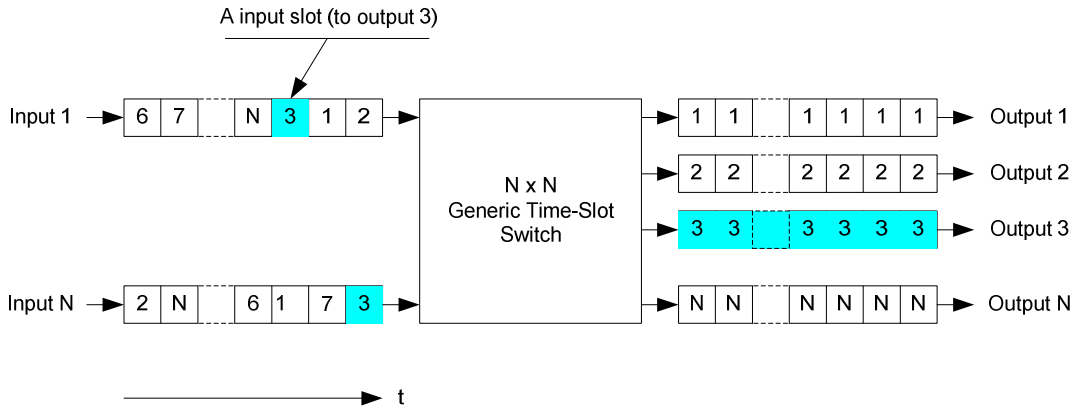


Figure 22 NxN time-slot switch

Assume the length of a packet is fixed; a NxN switch must support at least two following functions:

- Routing each packet to its destination output.
- Resolve the situation that two or more simultaneous packets arriving seek access to the same output.

4.4.2.1 Packet Switch with Queuing

Following is the analysis of lost packet performance in the absence of smoothing buffer.

In Figure 23, probability that the input has a packet arrived is p (p is also called the offered load or input fill factor). Probability of exact K input cells bound [61] for the same output is:

$$P_k(K=k) = \binom{N}{k}\left(\frac{p}{N}\right)^k \left(1-\frac{p}{N}\right)^{N-k} \tag{22}$$

where $$\binom{N}{k} = {}_N C_k = \frac{N!}{k!(N-k)!} \tag{23}$$

and K = 0,1…,N

Average number of loss packet L for a given output per a time slot is

$L = p + \left(1 - \frac{p}{N}\right)^N - 1$ and we obtain:

$\lim_{N\to\infty} L = p - 1 + e^{-p}$

Thus, output fill factor F could be found as:

$L = p - L = 1 - \left(1 - \frac{p}{N}\right)^N$ and we obtain:

$$\lim_{N\to\infty} F = p - e^{-p} \tag{24}$$

The fraction of incoming cells that is lost by the switch F_L is

$$F_L = \frac{L}{p} = 1 - \frac{1}{p} + \frac{1}{p}(1 - \frac{p}{N})^N \tag{25}$$

With larger switch, the loss is more significant.

4.4.2.2 Input Queuing and Traffic-handling Capability of an Input-Queued Packet Switch

Figure 23 is an input queuing switch block diagram. Input FIFO (First In- First Out) buffer is used to reduce the packet loss. The delay could be held up to (N)xT where T is the time slot length. To reduce the delay time caused by waiting for "up-front" cell (so-called blocking), a queued smoothing switch is utilized as shown in Figure 24. Each input has m buffers. Therefore the time slot switch now is not NxN but NmxNm

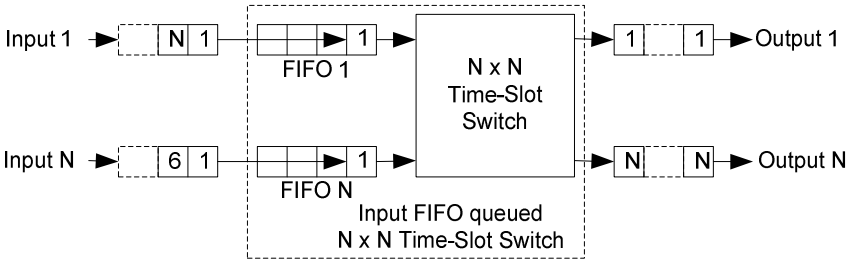


Figure 23 Input queueing switch

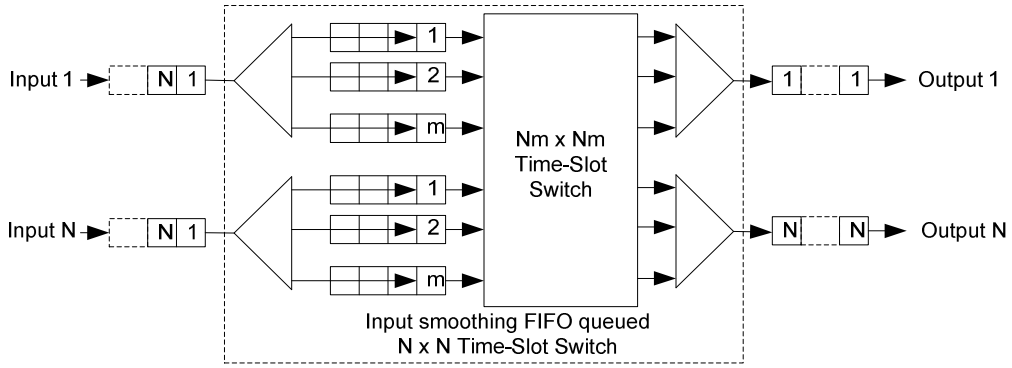


Figure 24 Input smoothing queued switch

The probability of a output to be filled P_F is:

$$P_F = \sum_{k-1}^{N} \binom{N}{k} \left(\frac{1}{N}\right)^k \left(1-\frac{1}{N}\right)^{N-k} = 1-\left(1-\frac{1}{N}\right)^N \tag{26}$$

$$\lim_{N\to\infty} P_F = 1-\frac{1}{e} = 63\%$$

For large *N*, only about 63% of the output time slots are filled. That is the maximum load for a packet switch [62].

4.4.2.3 Output Queuing

At the other end of the switch, the output could be in saturation, whereby the output would have to wait until the link is available. Unlike input queuing which only need *N* queues, an output queuing needs N^2 queues.

Mean Delay for a Packet Switch with Output Queuing is described as following:

$$\overline{D} = T\left(1+\frac{p(1-\frac{1}{N})}{2(1-p)}\right) \tag{27}$$

where *T* is time slot length.

When *N = 1* the delay time is equal to a time slot length.

4.4.3 Packet Loss

For whatever reason, if voice frame is not ready to play after playout delay time is ended, the packet that carries this frame is considered as lost. Packet latency exceeding the maximum delay threshold or packet that has been routed to the wrong destination are the most common packet loss situations.

4.5 Delay Jitter Measurement

4.5.1 Overview

Transmission delay jitter is the variance of transmitter delay [63]. Delay jitter causes the packets not to arrive after the same durations as they were sent. Packet delay and delay jitter at the receiver end (far-end) is used for quality monitoring and improvement. A practical method that allows measuring of the approximately delay jitter from far-end of a streaming packetized communication channel without packet timestamp has been introduced.

4.5.2 Delay Jitter in Packetized Communication

Assuming a streaming packetized communication channel, from packet start at the transmitter side to packet arrival at the receiver side and ready to playout, has a total transmission delay which could be simply expressed as:

$$D = D_{const} + l + D_{var} \tag{28}$$

where

- D_{const} is a minimum delay caused by any of sampling, coding, packetizing, queuing, propagation, and not subject to change from one packet to another [34].
- l is the packet interval (frame length).

- D_{var} *is* delay variance or Delay Jitter or Excess Delay or Jitter that could be different for each packet, depending on traffic congestion, switching route, etc. and subject to change from one packet to another.

Playout delay had been used to reduce the impact of jitter. Delay jitter is the base of playout delay buffer.

4.5.3 Delay Jitter Measurement for Packet without Timestamp

There will be no issues with finding transmission delay if every packet has a timestamp at both transmitter and receiver sides. However, this is not practical for packetized voice. Although an IP transmission delay could be estimated with a synchronized packet which has timestamp [7,8,64-65], voice packet may not have a timestamp on it. We have tried to assess the jitter without knowing what time packet was sent and with or without synchronizing packet timestamp. Other information we need is the packet number and frame length of each packet which is typically included in the voice packet.

Let us assume a set of $n+1$ voice packets has arrived. This set gives us a set t consisting of each packet arriving times and a set L consisting of each packet frame lengths:

$t= \{t_i\}$, $i = 1,..,n+1$

$L= \{l_i\}$, $i = 1,..,n+1$

Ideally, packet number i will arrive after a duration of $l_{i-1.}$

From (28) we have the first packet arriving time is:

$t_1=t_0 +D_{const} + l_1+J_1$

where t_0 is the time first voice packet start and J_1 is packet 1 delay jitter.

Packet 2 arriving time is:

$t_2 = t_0 + D_{const} + l_1 + l_2 + J_2$

where t_0 is the time first voice packet start and J_2 is packet 2 delay jitter.

In general, we have:

$$t_i = t_0 + D_{const} + \sum_{k=1}^{i} l_k + J_i \quad (29)$$

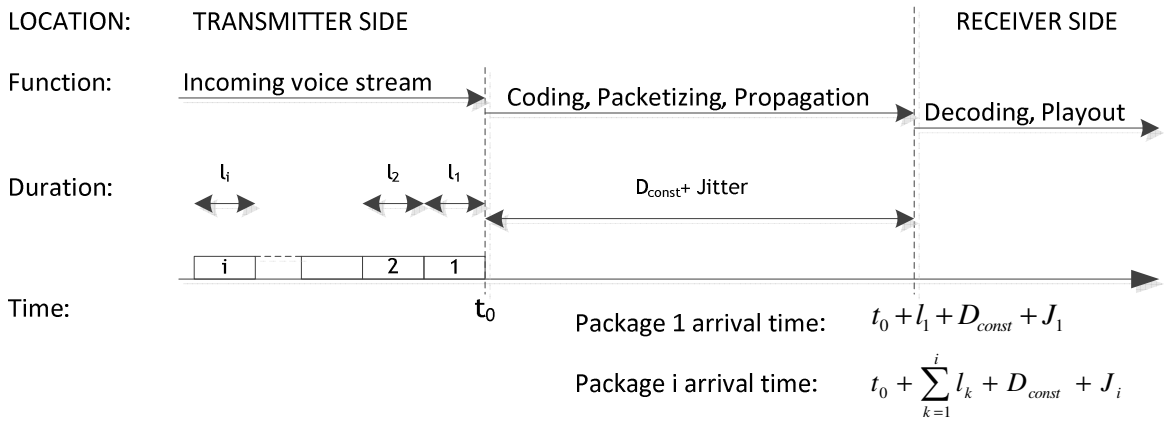


Figure 25 Packet voice arriving time

Removing frame lengths offset we obtain a substitute arrival time *t'*:

$$t'_i = t_i - \sum_{k=1}^{i} l_k = t_0 + D_{const} + J_i \quad (30)$$

Without loss of generality, we assume the minimum jitter is zero, or smallest value of *t'* will be as follow:

Minimum (t_i') = $m = t_0 + D_{const} + 0$

Thus we find the jitter set J by removing the offset m from (30).

$$J_i = t'_i - m \quad (31)$$

Following are some special situations that could happen and the solution for each situation.

The loss of packet could lead to missing of one of the frame length value delays, and equations (29), (30) and (31) will not have a solution. Since jitter J is not a negative value, we suggest using minimum frame length for the lost packets.

In many applications, the Codec uses a preset constant frame length. Total frame lengths offset in equations (29) and (20) could be written as:

$$\sum_{k=1}^{i} l_k = i \, . \, l \tag{32}$$

where l is the Codec frame length.

Most common application of using Jitter value is to decide the length of playout buffer. Depending on the application scheme, the Jitter could be quantized to reduce the computation complexity. In VoIP application, the quantized interval could be a minimum frame length.

We have used this method for a VoIP client application. The quantized jitter was used in a Markov model for playout delay buffer sizing.

4.6 Playout Delay and Markov Model

In the packet voice application, playout delay (POD) is used to reduce the number of packets loss caused by delay jitter. Longer POD will reduce the delay jitter loss; however, it will also degrade the quality [7, 66]. Thus, an adaptive delay model is used to optimize the playout delay. A practical adaptive POD based on Markov model has been introduced.

4.6.1 Delay Jitter in VoIP

In VoIP, if the frame length *l* is a constant, the total end to end delay (28) can be simply expressed as:

$$D = D_{const} + D_{var} \tag{33}$$

where D_{const} is a minimum delay caused by sampling interval (frame length), coding, packetizing, queuing, propagation, un-packetizing, decoding, assuming this will be the same for all packets.

D_{var} *is* delay variable or Delay Jitter or Excess Delay or Jitter that could be difference for each packet, depend on traffic congestion, switching route, etc.

The change of D_{var} could cause the packets not to arrive in the same arrangement as when they were sent out.

An example of no playout delay is shown in Figure 26, where the playout rule is simple as first in – first out (FIFO) and no delay. Assume that the frame length is *T*, any received packet will be played out immediately and the buffer size is infinity. The third packet and the fifth had some delay jitters which cause an extra delay of arrival. Although the third packet jitter is smaller than *T*, a part of the third packet could not be played without a gap between the second and the third packet and an overlap at the beginning of the fourth packet. If the fifth packet jitter is greater than a frame length T, it will not be played out and will be considered as a loss.

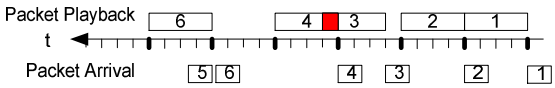


Figure 26 Packet loss when no delay plays-out or jitter buffer

Figure 27 illustrates how all packets will play out as they were sent for this example of packet arrival and with the play out delay greater than the maximum D_{var}.

In order for all arriving packets to play out in the right order, the delay should be greater than the maximum D_{var}. However, a longer delay will reduce the voice quality. An adaptive jitter buffer will optimize the delay, thus, optimizing the QoS. The jitter buffer size is adjusted during non-talk spurt periods.

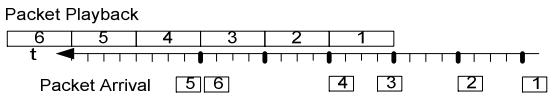


Figure 27 Play out with a delay or jitter buffer greater than maximum delay jitter

4.6.2 Basics of Fixed and Adaptive Jitter Buffer Models

The common schemes of playout delay in VoIP are to set up a jitter buffer length based on packet delay history. Typical fixed jitter buffer technique uses a statistical model, and adaptive

jitter buffer technique uses self-learning model. A fixed jitter buffer could use an average delay jitter or a delay that is set by a packet loss threshold.

The adaptive model use the jitter history (statically) to predict upcoming jitter and adjust buffer size accordingly. Markov [67-68] model for jitter prediction has been proposed. However, the jitter will need to be rounded up to an integer number in order to be used as the Markov's states. In addition, jitter will be quantized to simplify the computation of Markov model [4].

From the results of our test, the smallest voice frame length T is recommended as the quantized interval for Markov model, since delaying one voice frame could not impact too much on the voice quality. The typical delay buffer dealing with the large jitter range could be over 200 ms. For example, for a voice packet length of 20 ms and maximum jitter of less than 400 ms will have N = 20 states, or the transition matrix would be 20x20 only. If we use traditional jitter measurement with 1 ms quantization, then the number of states would be 400, and the transition matrix would be 400x400 instead.

A jitter delay t_n will be quantized with interval T equal smallest Codec frame length and its state i is:

$$i = INT(\frac{D_{var}}{T}) \tag{34}$$

where function *INT* is identified as following:

$$i - 1 \leq \frac{D_{var}}{T} < i \, , i = \{1,2, \dots , n\} \tag{35}$$

where T is minimum of Codec frame length.

We can now obtain a set of states (sample space), S = {1,2,3…n).

Let us call:

$$N = max\ (J_k),\ k = 1,2,..,n. \tag{36}$$

The J_k set will be used in the rest of the calculation.

4.7 Fixed Jitter Buffer Application

The packet which has jitter greater than playout delay will be considered as a lost packet. If the playout delay is set as *N,* there will be no packet loss by jitter. However, the maximum delay jitter could be very large. The voice quality will not be degraded significantly because of just one or two packet loss. If the threshold for packet loss is identified as *m* (%) then we could find the jitter buffer length *L* that satisfies that total percentage of delay jitter that greater than *L* is lesser than *m*:

$$\sum_{k=L}^{N} \sum_{i=1}^{N} p_{ik} \leq m \tag{37}$$

or

$$\frac{number\ of\ states\ grater\ than\ L}{n} \leq m \tag{38}$$

Eventually $1 < L \leq N$

The playout delay buffer therefore will be *L* instead of *N*.

Typically, we choose $m = 1\%$.

4.8 Self-learning, Adaptive Markov Model Application

The Markov model has been studied recently [10, 69, 70]. However, no report has been made on whether it has been tested or not. The research task is to investigate whether the method is realistic and how good it is. The research task is also to find its practical application.

We can establish a transition probability matrix, which shows the likelihood of the move from one state to another in the next step based on Markov. Figure 28 illustrates the probability of transition from state *i* to state *j* up to three steps.

$$P = \begin{array}{c} \\ 1 \\ \vdots \\ N \end{array} \begin{array}{c} \begin{array}{ccc} 1 & \dots & N \end{array} \\ \begin{pmatrix} p_{11} & p_{1..} & p_{1N} \\ p_{..1} & \ddots & p_{..N} \\ p_{N1} & p_{N..} & p_{NN} \end{pmatrix} \end{array} \tag{39}$$

where

$$p_{ij} = \frac{numbers\ of\ state\ i\ to\ state\ j}{numbers\ of\ state\ i} \tag{40}$$

For the next two steps we have:

$$p_{ij}^{(2)} = \sum_{k=1}^{N} p_{ik} p_{kj} \tag{41}$$

For the next three steps we have:

$$p_{ij}^{(3)} = \sum_{k=1}^{N} p_{ik} \sum_{m=1}^{N} p_{km} p_{mj} \tag{42}$$

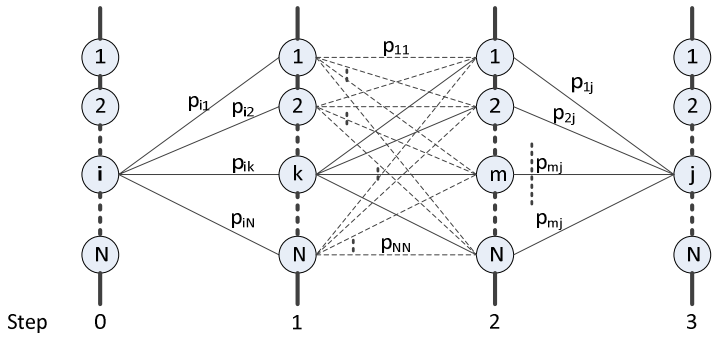


Figure 28 Markov model with *N* states and 3 steps, the probability of transition from state *i* to state *j*

And so on.

An adaptive model updates new jitter states continuously. The Markov model provides a prediction of the most effective delay play length and the buffer size adjustment will occur at the last talk spurt packet.

4.9 Experimental Result for a Simple Playout Delay Scheme

A simple scheme is shown in Figure 29. The jitter measurement is as proposed in part 4.5.3. The Markov model is the major part of delay control block. The Markov model provides a prediction at the last talk spurt packet. The delay control also makes changes to the jitter buffer size. Default input will include the length of the first Markov data set and will allow using fixed or adaptive buffer size.

An experimental scheme has been deployed in a voice client account in Tampa, FL where the other end is in Los Angeles, CA. The voice Codec packet length is 20ms. A set of 2000 voice packets was collected. Figures 30 is the plot of raw transmission delay jitter of all 2000 packets. It shows that these jitters are very bursty. Figure 30 also illustrates the percentage of delay and delay distribution. It shows that the maximum jitter is less than 200 ms. Figure 31 is the plot of the quantized values of the same data set. As we can see, the delay jitter is very bursty as well. Therefore, without playout delay buffer, there will be significant packet loss resulting in loss of voice quality.

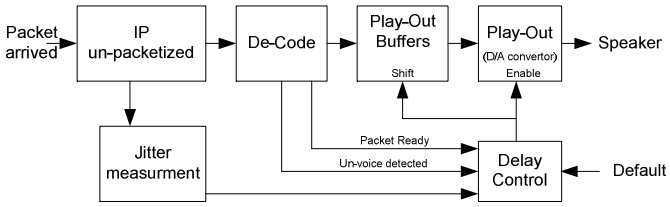


Figure 29 A simple playout delay scheme using Markov model

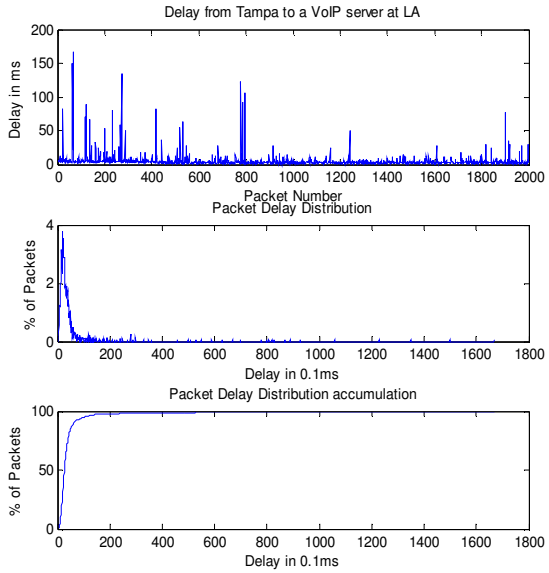


Figure 30 Delay of a voice channel from Tampa to Los Angeles

Figure 32 is the transition matrix (after digitized) with two steps. Since the maximum jitter is 10T, the Markov matrix is limited to 10^{th} order. The p_{ij} is showing the probability of the jitter length will be *j.T* (second) after *3.T* (second) if current jitter is *i.T* (second). Visually we can see a *2T* buffer size will be adequate.

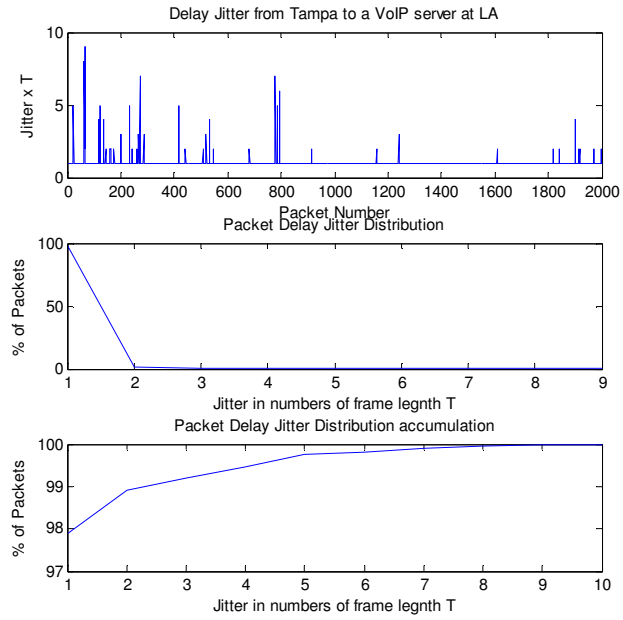


Figure 31 Quantized delay of a voice channel from Tampa to Los Angeles

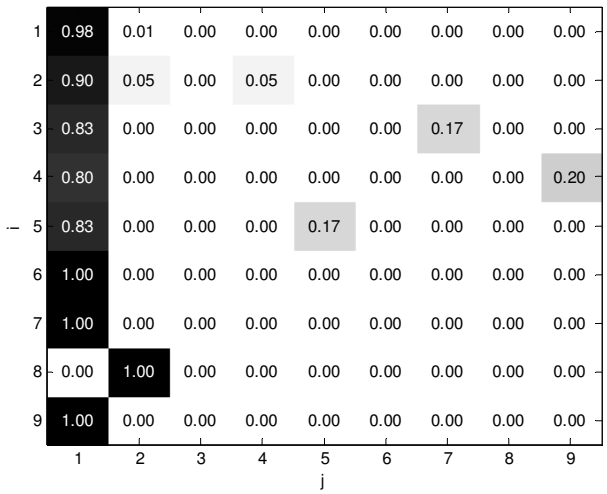


Figure 32 Transition matrix using two steps Markov model

There are few proposed playout delay buffer calculations [7],[8],[9], [10] which have been applied from theory to practice. Typically, the Markov model will be applied to the client side, which could be a small add-in application (app).

Applying two steps Markov model with limited states (ten), the computation complexity has been reduced significantly (at least 90%). The playout delay is reduced by 40% compared to fixed delay play out at the same packet loss rate, and the packet loss rate is reduced by 20% at the same average fixed playout delay.

4.10 Playout Delay Decision and Analysis Based on Markov Model

The Markov model uses the past data for future prediction. Using a large data set could lead to inaccurate prediction because the only up to date data could be used for the prediction due to the random change of network condition. On the other hand, using small data set will reduce the accuracy as well. Our experiment shows two seconds is sufficient. Expired data could become noise and should be removed.

Per Markov, the expected next jitter state J based on current jitter state i will be as follows:

$$J = \sum_{j=1}^{N} j p_{ij} \tag{43}$$

There were few issues with using the Markov model. The first problem is if $p_{ij} = 0$. That is the state in which i only occurred once at the end. In that case, there will be no solution for the next predicted jitter. However, in the playout delay control, there will be a solution. One of these solutions is to keep the latest playout delay.

The second issue of using Markov model is similar to the first one. When there is no state i in Markov model, the equation (40) will become infinity. In this case, the p_{ij} will be assigned to zero.

The playout delay could be based on one of following schemes:

- Next predicted jitter is j where p_{ij} is highest (j = 1 to N).
- Next predicted jitter is $\sum_{j=1}^{N} jp_{ij}$ per (43).

Since we always have a minimum playout delay offset at T or one minimum frame length we could use the round-up prediction per (43) as an additional delay.

4.11 Playout Delay Based on Markov Model Experiments

The playout delay using Markov model at first step has been tested with the same data set mentioned in the previous section. First 100 samples have been used as initial Markov model input for the first transition matrix. The predicted jitter number 101 is computed based on (43) and rounded up. The transition matrix is updated and used for the next jitters. Figure 33 demonstrates the result for the first step model and 100 predicted jitter values compared with the actual measured values. The Root Mean Square Error (RMSE) is from 0.022 to 0.028

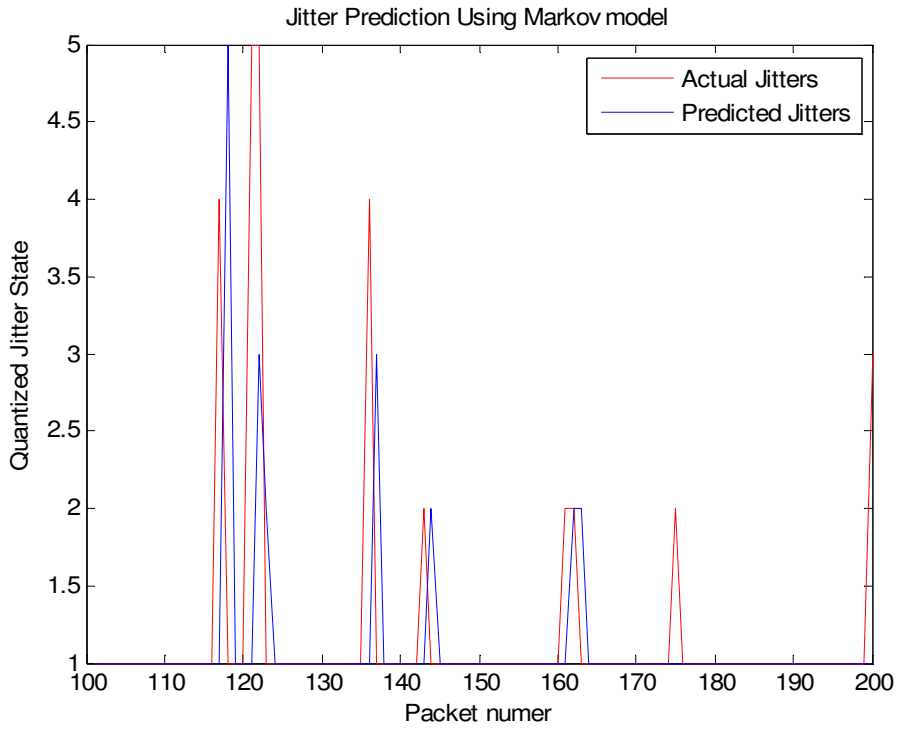


Figure 33 Jitter prediction using Markov model, first step

The playout delay using Markov model at second step has also been tested. Figure 34 demonstrates the result for the same data set. The Root Mean Square Error (RMSE) is from 0.024 to 0.035. The results show some inaccurate prediction at the large jitter change. As mentioned, this could be due to the random network condition change.

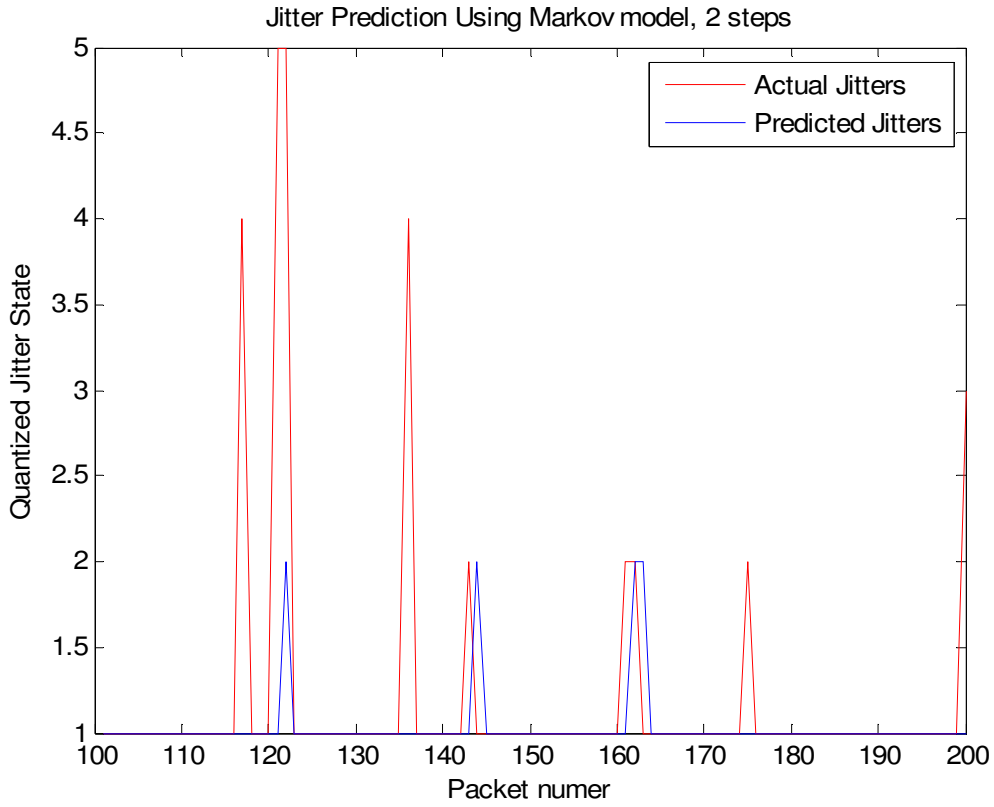


Figure 34 Jitter prediction using Markov model, second step

In another experiment, we added a gain on (43), i.e.,

$$J = g \sum_{j=1}^{N} jPij \tag{44}$$

where g is the gain, practically from 1 to 2.

The same experiment with g = 2 gave a result as shown in Figure 34. The Root Mean Square Error (RMSE) is from 0.05 to 0.055. However, the results show that there is no packet loss with Markov prediction and the average playout delay is less than maximum delay jitter.

Repeating the test with larger data, jitter prediction gets more accurate with the first step Markov model. In conclusion, using Markov prediction model, the previous jitter states should be collected up to date and should be large enough to compute a Markov transition matrix. The Markov transition matrix could contain some infinity or unidentified value if the current state J_n has not occurred before. The solution is to assume $J_n = J_{n+1}$. The first step model gave the highest accuracy compared with the higher step models. Adding gain for predicted jitter calculation could increase the playout delay within a small margin, however, will reduce the packet loss ratio. The Markov adaptive playout delay has been tested and seems to improve the quality of the channel that has an aggressive jitter change.

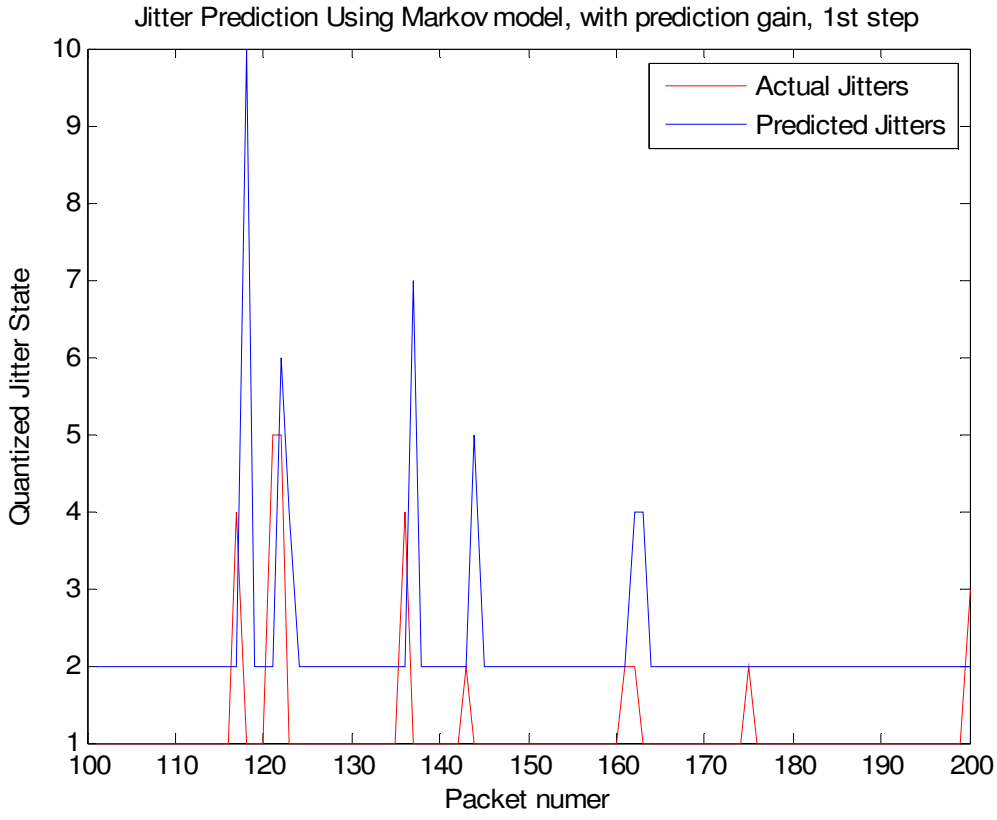


Figure 35 Jitter prediction using Markov model, first step with gain=2

4.12 Kalman Filter and Jitter Prediction Improvement

Recently, there have been a number of studies done on using Kalman filter for packet loss improvement [11, 12,69-71]. The Kalman filter works as a tracking mechanism, improving the next prediction based on previous prediction and measurement. The Kalman filter is used to reduce the impact of noise on the input. The Kalman filtering is also used for delay compensation (playout delay). Sirirat et al. [13] have proposed a Kalman filtering estimation. The linear dynamic process is represented by a jitter model assuming a Gauss-Markov process autocorrelation. Comparing the results of H. Bi et al. [71] with our results for the Markov model, it seems to be the same with Root Mean Square Error (RMSE) from 0.022 to 0.035.

Kalman filter could be used with Markov prediction, where the Markov model will provide the first prediction and the Kalman filter will provide updated prediction after receiving new measurement. For future work, the study with the assumption that jitter is a Gaussian distribution should be carried out as well.

Table 4 is a comparison of the improvement of R factor based on the same jitter data set (2000 samples), between playout delay using adaptive Markov models, packet loss threshold and fixed average jitter. The adaptive model gives the same packet loss ratio with packet loss threshold model. However, average playout delay reduced significantly while the average jitter model has more packet loss.

Table 4 Jitter playout delay improvement under R-factor

	Adaptive Markov model	Packet loss threshold model	Average jitter model (Benchmark)
Theoretically performance	N/A	N/A	N/A
Empirical performance	88	80	65

CHAPTER 5: SUMMARY AND SUGGESTION FOR FUTURE RESEARCH

5.1 Summary

The VoIP quality is dependent on the network condition. Planning with a minimum bandwidth is the first step of quality control. This involves changing the Codec, compressing or simplifying IP header to reduce bandwidth in order to prevent traffic congestion and reduce the impacts of packet loss. This research has analyzed the VoIP planning based on E-model and its factors and has provided a simplified calculation for VoIP bandwidth.

A Method for calculating a limited duration (or a section) delay jitter without timestamp of each packet and with variable Codec frame lengths has been proposed in this research. This will be very useful in case the RTP header is being removed, which could close the gap in the engineering solution for jitter delay prediction based on Markov model.

In order to deal with jitter as the major impact on network impairment, playout delay has been used widely. The research has focused on optimized playout delay to improve the VoIP quality. An approach using Markov prediction and quantized jitter has been proposed and tested. The research has pointed out an important infinity loop in Markov model that has not been addressed before. The research also included extensive tests and has confirmed that Markov has 8% improvement more than fixed threshold method and 28% more than average jitter method. The Markov model seems to work better at the first step and could be more efficient if a prediction gain is applied.

Future research could focus on Kalman filter and Markov autocorrelation models. Due to the growth of the social network and smart grid, a social network study on the probability of a voice call on the social network and the role of voice communication in smart grid also need further study. Applying complex network on VoIP could be an interesting research area too.

5.2 A Maxell Model for Packet Loss Caused by Jitter

A packet loss is considered as a blocked packet during transportation. Dr. Thompson in his book [72] stated that in Maxell model, a flux F at receiver sphere at Δt is zero for a blocked stream. By increasing Δt F is greater than zero.

$$F = \oint_t^{t+\Delta t} K dt \tag{45}$$

where F is total packet received and K is receiver throughput.

In the future, research on all factors that relate to the flux could be an interesting road map. Since not all factors are independent, i.e., Codec and jitter in combination could gain more effort to block the packet.

5.3 VoIP and Social Network

The possibility of allowing a voice call from and to a social network is still under investigation. The regulator and engineers should work closer to prepare a standard not just for the voice quality but also for security. Today a telephone number could be an identity number while a social network account is not. As soon as a social network account becomes an identity number, telephony could be extended to the social network, and the regulation and standard will be changed at that time to ensure improved quality.

5.4 Complex Network and VoIP

Recently, the development of Complex Network gives us another approach for VoIP study. In the VoIP Complex Network, each node could be a hop and connection could be a link. Using

the node and its link as a property set, we could use a graph to simulate the VoIP behaviors and estimate the quality of a call between two nodes [73].

5.5 Voice in Smart Grid

If voice becomes an add-in feature of a communication network, then there is a high potential that VoIP will be embedded in the Smart Grid communication. Recently there are few papers addressing this issue. Along with smart control and monitoring, smart voice communication is also mentioned. The voice in Smart Grid deals with higher priority data communication to keep its quality.

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